THE JOURNAL of the Acoustical Society of America

• CONTENTS •

Finite Acoustic Filters . . . . . R. B. LINDSAY 211

Transposition of Speech Sounds . . . . . ERNST BÁRÁNY 217

Detection and Location of Laminations in Steel Plates . HARVEY C. HAYES 220

Vibrating String Considered as an Electrical Transmission Line . . . . . WINSTON E. KOCK 227

Direct Recording and Reproducing Materials for Disk Recording . . . . . A. C. KELLER 234

Principles Underlying the Tuning of Keyboard Instruments to Equal Temperament . . . . . G. F. HERRENDEN HARKER 243

Sound Analysis . . . . . HARRY H. HALL 257

Current Publications on Acoustics . . . . . F. A. FIRESTONE 263

Acoustical Society News 267

Index to Volume 8 268

Vol. 8 APRIL, 1937 No. 4

THE JOURNAL
OF THE
ACOUSTICAL SOCIETY OF AMERICA

Volume 8 Number 4

APRIL • 1937

Finite Acoustic Filters

R. B. LINDSAY
Brown University, Providence, Rhode Island

(Received October 27, 1936)

Most of the previous theoretical work on acoustic filtration has dealt with idealized infinite structures. The present paper is a theoretical study of finite filters such as are used in practice. Considering an acoustic structure with air as the medium with main line impedance Z and with n branch impedances Z_b (pure reactances) equally spaced at intervals $2l$, it develops that one can express the power transmission ratio for the structure in the simple form

$$P_r = 1 \Big/ \left(1 + \frac{Z^2}{4|Z_b|^2} \cdot \frac{\sin^2 nW}{\sin^2 W}\right), \qquad (1)$$

where $$\cos W = \cos 2kl + iZ/2Z_b \cdot \sin 2kl, \qquad (2)$$

the familiar parameter of the transmission theory of filtration (reference 2). The plot of P_r in (1) as a function of frequency shows the presence of transmission and attenuation bands which approach those of the corresponding ideal infinite case as n becomes large compared with unity. Comparison of (1) with the actual transmission measurements of Stewart (reference 2, pp. 170, 175) shows satisfactory agreement.

The phase change which occurs between successive sections in a filter is also computed and it is shown that unlike the situation in the infinite case the phase change is a function of the section. However, at relatively high frequencies and for large n this approximates to the total phase change along the structure divided by n. Comparison with such experimental results as are at present available yields reasonably satisfactory agreement.

FINITE ACOUSTIC FILTERS

ACOUSTIC filtration in fluids and solids has been studied extensively during the past fifteen years from both experimental and theoretical standpoints.[1] In most cases the theory has been presented from the standpoint of an infinitely recurrent structure, though the results have been applied with some success to the finite structures experimentally investigated.[1A] In particular the frequency limits of the

[1] For an account of the work up to 1930 the reader may consult: G. W. Stewart and R. B. Lindsay, *Acoustics* (D. Van Nostrand, N. Y., 1930), Chap. VII. For a list of more recent papers dealing particularly with filtration in solids, see J. Acous. Soc. Am. **8**, 42 (1936). Mention of recent European literature will be found in Trendelenburg, *Klänge und Geräusche* (Berlin, 1935), pp. 24–25. It is worth while recording here that more recent search of the literature has brought to light an old paper by Horace Lamb "On Waves in a Medium Having a Periodic Discontinuity of Structure" (Memoirs and Proceedings of the Manchester Literary and Philosophical Society **42**, 1 (1897–1898)). This discusses filtration of waves in a loaded string. Acoustic filtration appears also to have been studied by N. Kasterin in some papers on the propagation of waves in media with discontinuities as well as lamellar media (Amst. Ak. Versl. **6**, 460 (1898) and Arch. Néerland. **5**, 506 (1900)).

[1A] The effect of inserting a finite filter structure in an acoustic line has indeed been studied thoroughly by W. P.

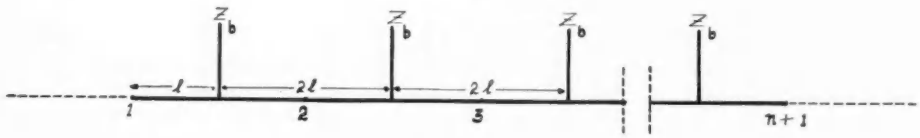


FIG. 1.

transmission and attenuation bands predicted by theory have been approximately verified in the cases so far tested by experiment. Up to the present, however, little or no attempt has been made to compare the theoretical calculations of the transmission through a finite filter with actual experimental measurements. It is the purpose of the present paper to examine this question.

We shall first consider the system indicated schematically in Fig. 1. This consists of an acoustic conduit of cross-sectional area S with n branch lines at equal intervals of length $2l$. The acoustic impedance of each branch at the junction is Z_b and is assumed to be a pure reactance. The portion of the structure from a point midway between two branches to the next successive midpoint constitutes one section of the structure (e.g., from 2 to 3 in Fig. 1). The whole structure of n sections is assumed to be inserted in an infinite acoustic line. Acoustic radiation in the form of a plane harmonic wave coming from the left is incident on the structure at 1 and leaves it at $n+1$ on the right in the form of a plane wave progressing solely to the right, the possibility of reflection being ruled out by the infinite line. We shall suppose that the infinite line is a regular acoustical conduit of cross-sectional area S_i on the left and S_t on the right of the filter structure.

Following the usual procedure and employing the proper boundary conditions[2] we readily obtain the following equations connecting the excess pressure and volume current at the midpoints j and $j+1$ (j being any integer between 1 and n)

$$p_{j+1}=Ap_j-iBZ\dot{X}_j, \tag{1}$$

$$\dot{X}_{j+1}=A\dot{X}_j-iC/Z\cdot p_j, \tag{2}$$

wherein
$$\begin{aligned} A&=\cos 2kl+iZ/2Z_b\cdot\sin 2kl,\\ B&=\sin 2kl+iZ/Z_b\cdot\sin^2 kl,\\ C&=\sin 2kl-iZ/Z_b\cdot\cos^2 kl. \end{aligned} \tag{3}$$

In these expressions $Z=\rho_0 c/S$ is the acoustical impedance of a plane harmonic progressive wave. The mean density of air is ρ_0 and the velocity of sound in air is c. As usual $k=2\pi\nu/c$, where ν is the frequency.

A brief examination of (3) indicates that there exists the relation

$$A^2+BC=1.$$

Consequently, if we call $A=\cos W$, we can write $B=(B/C)^{\frac{1}{2}}\cdot\sin W$ and $C=(C/B)^{\frac{1}{2}}\sin W$. Eqs. (1) and (2) may now be written for the case $j=n$

$$p_{n+1}=p_n\cos W-iZ\dot{X}_n(B/C)^{\frac{1}{2}}\cdot\sin W, \tag{4}$$

$$\dot{X}_{n+1}=\dot{X}_n\cos W-i/Z\cdot p_n(C/B)^{\frac{1}{2}}\cdot\sin W. \tag{5}$$

It is now easy to establish by mathematical induction the corresponding relations connecting p_{n+1} and $\dot{X}_{n+1}$ with p_1 and $\dot{X}_1$. These prove to be

$$p_{n+1}=p_1\cos nW-iZ\dot{X}_1(B/C)^{\frac{1}{2}}\cdot\sin nW, \tag{6}$$

$$\dot{X}_{n+1}=\dot{X}_1\cos nW-i/Z\cdot p_1(C/B)^{\frac{1}{2}}\cdot\sin nW. \tag{7}$$

Now by the boundary conditions at $n+1$, $\dot{X}_{n+1}$ is the volume current and p_{n+1} the excess pressure in the wave traveling out of the terminus of the structure. To get the transmission through the structure we must compare these with the corresponding quantities for the wave incident on the structure at 1. Let us denote the excess pressure in the incident wave to the left of 1 (chosen as the origin of coordinates) by p, where

$$p=p_0e^{-ikx}+p_0'e^{ikx}. \tag{8}$$

If correspondingly $\dot{X}$ is the volume current in the incident wave, we have

$$\dot{X}=(S_i/\rho_0 c)(p_0e^{-ikx}-p_0'e^{ikx}). \tag{9}$$

Here p_0 is the excess pressure in the *incident*

Mason (Bell Sys. Tech. J., **6**, 258 (1927)). The method of presentation differs from that of the present paper and, moreover, no attempt is made at a comparison between theoretical calculations and experimental measurements.

[2] Cf. Stewart and Lindsay, *Acoustics*, p. 334 ff.

wave at the origin and p_0' that in the *reflected* wave at the same place. We shall define as the power transmission ratio of the structure

$$P_r = Z_i |p_{n+1}|^2 / Z_t |p_0|^2 \tag{10}$$

with $Z_i = \rho_0 c/S_i$ and $Z_t = \rho_0 c/S_t$. The boundary conditions at the origin are

$$\begin{aligned} p_1 &= p_0 + p_0', \\ \dot{X}_1 &= S_i/\rho_0 c \cdot (p_0 - p_0'). \end{aligned} \tag{11}$$

Substituting for p_1 and $\dot{X}_1$ from (11) into (6) and (7) with the subsequent elimination of p_0' and recalling that we may write $\dot{X}_{n+1} = p_{n+1}/Z_t$, we have finally

$$\frac{p_{n+1}}{p_0} = \frac{2}{(1+Z_i/Z_t)\cos nW + i(Z_i/Z \cdot (C/B)^{\frac{1}{2}} + Z/Z_t \cdot (B/C)^{\frac{1}{2}}) \sin nW}. \tag{12}$$

For the transmission ratio we therefore have

$$P_r = Z_i/Z_t \cdot \frac{4}{(1+Z_i/Z_t)^2 \cos^2 nW + (Z_i/Z \cdot (C/B)^{\frac{1}{2}} + Z/Z_t \cdot (B/C)^{\frac{1}{2}})^2 \sin^2 nW}. \tag{13}$$

It should be emphasized that C/B can become negative. However, in this case, sin W must be pure imaginary, since B and C are always real. Hence sin nW will also be pure imaginary, while cos nW is always real, since A is real. This justifies our procedure in passing from (12) to (13) for all cases.

A useful alternative form for (13) is

$$P_r = Z_i/Z_t \cdot \frac{4}{(1+Z_i/Z_t)^2 \cos^2 nW + (Z_i/Z \cdot C + Z/Z_t \cdot B)^2 \sin^2 nW / \sin^2 W}. \tag{13'}$$

In the experimental arrangements to which we shall principally refer, $Z_i = Z = Z_t$ and (13′) reduces to the simpler form

$$P_r = \frac{1}{1 + Z^2/|Z_b|^2 \cdot (\sin^2 nW)/(4 \sin^2 W)}. \tag{14}$$

Now in the corresponding case of an infinitely recurrent structure it has been shown[2] that the structure transmits those frequencies for which $|\cos W| \leq 1$, while for $|\cos W| > 1$ attenuation results. Now when $|\cos W| > 1$ we have $W = W_r + iW_i$, where W_r and W_i are the real and imaginary components of W, respectively. Since cos W is real, $W_r = n'\pi$ where n' is an integer. Consequently $\cos nW = \pm \cosh nW_i$ and $\sin nW = \pm i \sinh nW_i$, while $\sin W = \pm i \sinh W_i$. It is clear from (14) that in this case as n increases the denominator will increase without limit and P_r will become very small. On the other hand for $|\cos W| \leq 1$, sin W and sin nW are both numerically less than or equal to unity for any n and the possibility exists for the second term in the denominator to become very small or P_r to become close to unity. This exhibits the close connection between the attenuation and transmission bands of the infinitely recurrent structure and the transmission ratio of the finite structure.

To see the connection in greater detail it is of interest to examine a case investigated experimentally by Stewart.[3] This is a low pass filter consisting of an acoustic line with equally spaced Helmholtz resonators. The following are the relevant data:

ρ_0 = equilibrium density of air = 12.05×10^{-4} gram/cm^3,
c = velocity of sound in air (20°C) = 3.44×10^4 cm/sec.,
S = area of main line = $9\pi/16$ cm^2,
$2l$ = 1.67 cm,
$Z_b = i(\omega M - 1/\omega C)$, where $M = \rho_0/c_0$, $C = V/\rho_0 c^2$, and c_0 = conductivity of orifice into resonator orifice = 2.26 cm,
V = volume of resonator chamber = 4.36 cm^3,
n = number of sections = 4.

Fig. 2 plots the theoretical curve for P_r obtained from Eq. (14) (the full line) and Stewart's experimental curve (the dashed line). As might have

[3] Cf. Stewart and Lindsay, *Acoustics*, p. 170. For more details cf. Phys. Rev. **20**, 546 (1922).

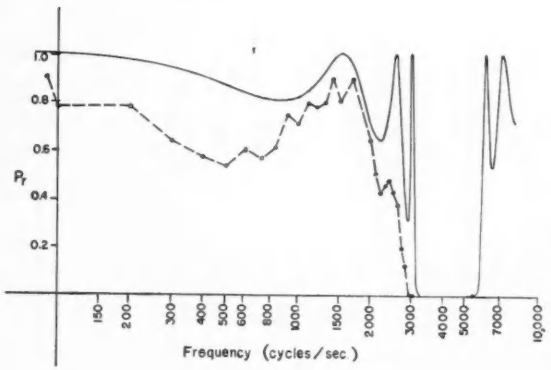


FIG. 2.

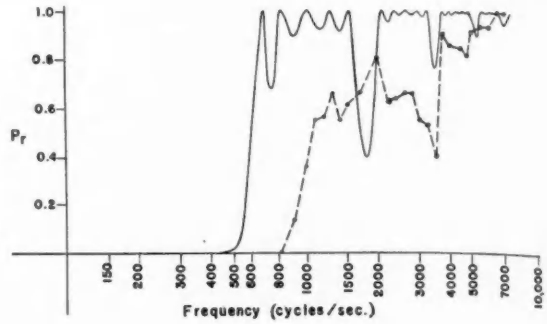


FIG. 3.

been expected the transmission is lower than that theoretically predicted since the theory takes no account of irreversible dissipation. However, it is very interesting to note that the general trend of the curves is the same in both theory and experiment. Stewart indeed was more interested in the cut-off frequency than in the actual transmission. His measuring device was unable to do more than barely detect the second transmission band beginning around 5700 cycles.

A similar illustration is provided by a high pass filter, described by Stewart,[4] consisting of an acoustic line with equally spaced orifices as branches. The dimensions follow

$2l=10$ cm; $n=6$,
$Z=56.09$ gram/cm^4 sec.,
$Z_b=i\rho_0\omega/c_0=0.896i\nu$ (c_0 = conductivity of orifices = 0.085 cm),
$Z/2Z_b=313/i\nu$.

Fig. 3 plots the theoretical (full line) and experimental (dashed line) curves for P_r. The general agreement is seen to be good considering the rather crude experimental method. It may be noted in passing that better agreement with experiment results if one uses for the conductivity $c_0=0.17$ cm or twice the value estimated by Stewart (see Fig. 4). The estimation of the conductivity of orifices remains a very uncertain factor in acoustical calculations. Particular attention should be paid to the fact that the attenuation regions occurring at high frequencies predicted merely by the examination of the behavior of cos W actually are not observed in the finite case due to the term $Z^2/|Z_b|^2$ in the expression for P_r. This term decreases as the frequency increases; consequently independently of n, $P_r\to 1$ as the frequency grows without limit. This is confirmed by all the experimental studies of Stewart on high pass filters.

The phase change which occurs in the passage of the sound from one section to the next in the filter is of some interest, as it also has been studied experimentally. For the sake of definiteness and simplicity we shall discuss the case where $Z=Z_i=Z_t$. In the first place the *total* change in phase involved in the passage of acoustic radiation through the whole structure is readily calculated from Eq. (12). Thus utilizing the special impedance relations just mentioned,

$$\frac{p_{n+1}}{p_0}=\frac{\dot{X}_{n+1}}{\dot{X}_0}=\frac{1}{\cos nW+(i/2)((C/B)^{\frac{1}{2}}+(B/C)^{\frac{1}{2}})\sin nW},$$

whence

$$\frac{p_{n+1}}{p_0}=\frac{|p_{n+1}|}{|p_0|}\cdot\frac{\cos nW-(i/2)((C/B)^{\frac{1}{2}}+(B/C)^{\frac{1}{2}})\sin nW}{(\cos^2 nW+\frac{1}{4}((C/B)^{\frac{1}{2}}+(B/C)^{\frac{1}{2}})^2\sin^2 nW)^{\frac{1}{2}}}.$$

Therefore if θ is the phase change

[4] Reference 2, p. 175.

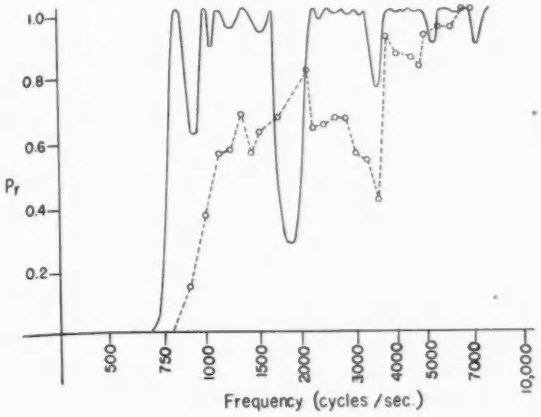


FIG. 4.

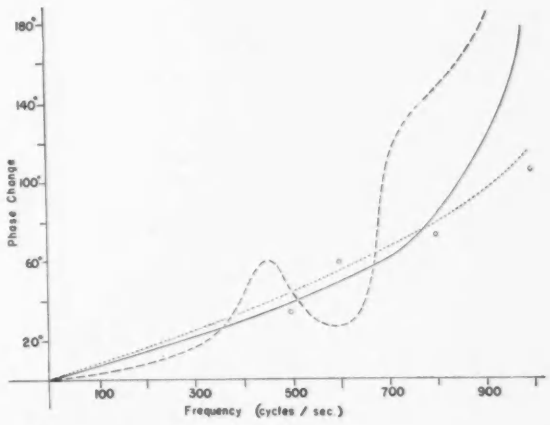


FIG. 5.

$$\cos\theta=\frac{\cos nW}{(\cos^2 nW+\tfrac{1}{4}((C/B)^{\frac{1}{2}}+(B/C)^{\frac{1}{2}})^2\sin^2 nW)^{\frac{1}{2}}}$$

and

$$\tan\theta=\frac{-((C/B)^{\frac{1}{2}}+(B/C)^{\frac{1}{2}})}{2}\tan nW. \tag{15}$$

Before we discuss this, let us calculate the change in phase involved in the passage from one section to the next, e.g., from the jth to the $(j+1)$st. Reverting to Eqs. (1) and (2) written in terms of $\sin W$ and $\cos W$ and utilizing the impedance relation $p_{n+1}=Z\dot{X}_{n+1}$, we can readily compute $p_n/\dot{X}_n$. Thus

$$p_n/\dot{X}_n=\frac{Z(\cos W+i(B/C)^{\frac{1}{2}}\sin W)}{(\cos W+i(C/B)^{\frac{1}{2}}\sin W)}.$$

It is then an easy matter to establish by mathematical induction the general relation (j any integer between 1 and n).

$$p_j/\dot{X}_j=\frac{Z[\cos(n-j+1)W+i(B/C)^{\frac{1}{2}}\cdot\sin(n-j+1)W]}{[\cos(n-j+1)W+i(C/B)^{\frac{1}{2}}\cdot\sin(n-j+1)W]}. \tag{16}$$

This brings out clearly the fact that the mid-section impedance is not in general equal to Z. However, from the form of B and C (Eq. (3)) it is clear that as the frequency increases, for both low and high pass filters, $B\to C$ and $p_j/\dot{X}_j\to Z$. Hence in the limit of high frequencies the mid-section impedance approaches the uniform value of Z for all sections.

If now we use (16) to write p_{j+1} in terms of $\dot{X}_{j+1}$ and substitute into Eqs. (1) and (2), eliminating p_j between the two, we get the general relation

$$\frac{\dot{X}_{j+1}}{\dot{X}_j}=\frac{\sin(n-j)W-i(B/C)^{\frac{1}{2}}\cos(n-j)W}{\sin(n-j+1)W-i(B/C)^{\frac{1}{2}}\cos(n-j+1)W}. \tag{17}$$

We may therefore write

$$\frac{\dot{X}_{j+1}}{\dot{X}_j}=\frac{|\dot{X}_{j+1}|}{|\dot{X}_j|}\cdot\frac{\begin{matrix}\{\sin(n-j)W\sin(n-j+1)W+B/C\cdot\cos(n-j)W\cos(n-j+1)W\}\\ +i(B/C)^{\frac{1}{2}}\cdot\{\cos(n-j+1)W\cdot\sin(n-j)W-\cos(n-j)W\sin(n-j+1)W\}\end{matrix}}{(\sin^2(n-j)W+B/C\cdot\cos^2(n-j)W)^{\frac{1}{2}}(\sin^2(n-j+1)W+B/C\cdot\cos^2(n-j+1)W)^{\frac{1}{2}}}.$$

This leads to the phase change θ in $\dot{X}$ in passing from the jth to the $(j+1)$st section in the form

$$\tan\theta=-\frac{B}{\sin(n-j)W\sin(n-j+1)W+B/C\cdot\cos(n-j)W\cdot\cos(n-j+1)W}. \tag{18}$$

It is interesting to observe that although the general phase change is a function of the section as well as the frequency, for high frequencies

$$\tan\theta\rightarrow-\tan W \tag{18'}$$

independent of the section. Now in the infinitely recurrent filter the phase change is precisely $-W$ when W is real, and the real part of W (an integral multiple of π or zero as we have seen above) when W is complex. The connection between the finite and infinite cases in the limit thus appears clear, as is indeed further evident in the case of the *total* phase change through the whole structure. Reverting to Eq. (15) we see that in the limiting case of high frequencies

$$\tan\theta\rightarrow-\tan nW. \tag{15'}$$

It is of interest to compare the application of Eqs. (15) and (18) to a special low pass filter studied by Stewart[5] with his experimental results. Fig. 5 presents the comparison. The discrete points represent the phase change *per section* experimentally measured as a function of the frequency within the first transmission region (cut-off around 1100 cycles). The dotted curve represents the theoretical value computed by Stewart's lumped impedance theory.[5] The full curve is given by evaluating θ from (15) and dividing by 10 since there are 10 sections in the filter and this may be expected to give the average phase change per section. The dashed curve is the result of plotting θ from Eq. (18) with $j=5$, i.e., the phase change for the middle section of the structure. The similarities and differences are obvious. It is extremely desirable that new and accurate phase measurements should be made to check these theoretical results.

The author wishes here to express his sincere appreciation of assistance in connection with arithmetical calculations and curve-plotting rendered at various times by Messrs. A. O. Williams, Jr., T. G. Barnes and R. L. Mooney, graduate students in the department of physics in Brown University.

[5] Stewart, Phys. Rev. **23**, 520 (1924).

Transposition of Speech Sounds

ERNST BÁRÁNY
University of Upsala, Sweden
(Received October 4, 1936)

EXPERIMENTS with transposition of speech sounds have a great theoretical interest because it is not yet clear whether the characters of the speech sounds are determined by the pitch of the reinforced regions of harmonics,[1] the formants, or by the intervals between these. Nor is it clear whether the intensity relations between the formants play any prominent part. If it is possible to transpose a speech sound without altering its fundamental nature of speech sound and its recognizability, the intervals between its components clearly are its most important character.

Transposition experiments with gramophone records rotating at different speeds have been published from several quarters[2–4] and have shown that at least some vowels may be changed one octave higher without losing much of their intelligibility. The consonants seem to be more sensitive, as may perhaps be concluded from the fact that Fletcher's curve of articulation, which is based on vowels and consonants declines more rapidly at higher speeds than those of Engelhardt and Gehrcke who investigated only vowels. Transposition of synthetic vowels is reported to be successful by Kucharski[5] the possible transposition sometimes being as great as 1.5 octaves upwards and downwards, but differing in a marked way between the different vowels. He gives no figures of the articulation obtained.

The transposition with the gramophone suffers by the frequency distortion introduced by the reproducing system and becoming worse with the grade of transposition. Further, the speed of the phonetic processes is altered. This should be of importance in the case of short consonants, where the time intensity characteristic of the ear[6] and perhaps the transients in the middle ear[7] begin to play a part.

During an investigation on a related subject the author observed a transposition of speech sounds when spoken with hydrogen in the airways instead of air. The hydrogen causes the cavity resonances to change upwards, at the best some 2 octaves and therewith the whole formant region is lifted, leaving only the fundamental of the vocal cords unchanged. The voice alters in a very marked way but remains clearly understandable, at least for connected speech. Whispering sounds very sharp but is not altered as dramatically as normal speech.

As I discovered later, this observation is not new. John Tyndall[8] in 1867 demonstrated the experiment during a lecture at the Royal Institution, without mentioning, however, its importance to the theory of speech sounds. He said about the voice in hydrogen that it was, "hollow, harsh and unearthy, I cannot otherwise describe it."

Tyndall attributed the change to the passing of the sound from a light gas into a heavier. It is difficult to say whether reflections occurring at the boundary of the hydrogen cloud really play a part. If one assumes the boundary to be a plane junction of two infinite media, an attenuation independent of frequency of 7 decibels would occur. As the junction is not plane and the hydrogen side is not infinite it becomes necessary to integrate velocity potentials, but as the boundaries are not known this has not been possible.

Personally, the author attributes the greatest part of the change to changes in the resonance frequencies of the formant determining cavities.

Further factors influencing the voice in hydrogen are the changes of the radiating properties

[1] Vierling, Ann. d. Physik **26**, 219 (1936).
[2] Stumpf, *Die Sprachlaute* (Springer, Berlin, 1926), p. 228. Here a discussion based on experiments with mechanical reproducers.
[3] Fletcher, *Speech and Hearing* (Macmillan, London, 1929), p. 292.
[4] Engelhardt und Gehrcke, Zeits. f. Psych. **115**, 1 (1930).
[5] Kucharski, Comptes rendus **195**, 979 (1932).
[6] v. Békésy, Physik. Zeits. **30**, 115 (1929).
[7] v. Békésy, Physik. Zeits. **34**, 577 (1933) and Akustische Zeits. **1**, 19 (1936).
[8] Tyndall, *Sound*. A Course of Eight Lectures Delivered at the Royal Institution of Great Britain. (Longmans, Green and Co., London, 1867), p. 9. Referred to in Scheminzky, *Die Welt des Schalles* (Wien, 1935).

of the vocal cords and the mouth and changes in the decrement of the cavities, which are likely to occur. There ought not to be any appreciable change in the vibration pattern of the vocal cords, because the loading of the cords by the gas is very slight.[9]

In order to get an idea of the articulation in hydrogen the author arranged an improvised articulation test in a living room, using one invisible caller and seven observers, all of the participating people being totally untrained for the purpose. Syllables were obtained by random combinations of the consonant-vowel-consonant pattern, the 27 most important Swedish speech sounds occurring with equal frequency. In order to eliminate the possibility of the caller altering his voice in order to correct the pronunciation in hydrogen he was provided with a stethoscope through which a masking noise could be introduced to his ears, masking his own voice. The reading of the syllables list was controlled by a separate observer with a duplicate of the list.

The filling of the airways with hydrogen was made by taking a deep breath from a rubber bag containing pure hydrogen, after a maximal expiration in free air. In order to ensure filling the nose with the right gas, a slight exhalation preceded the reading after each filling of the lungs. The mouth was directed downwards in order to prevent the hydrogen from escaping.

Control series with air and without noise were arranged. 99 syllables were read, 27 with hydrogen and masking noise, 27 with air and noise, 18 with hydrogen but without noise and 27 with air and without noise. Each sound thus was observed 35 times in hydrogen and 42 times in air. The results appear in Table I.

TABLE I.

	Articulation %		
	First consonants	Vowels	Last consonants
Hydrogen, noise	67	64	53
Air, noise	90	93	76
Hydrogen, silence	76	73	63
Air, silence	96	96	79
Articulation in hydrogen, divided by corresponding articulation in air			
noise	75	69	70
silence	79	76	80

The analysis has not been produced to single speech sounds, as the material is too spare and the variability too large. The only distinction made is between first and last consonants and vowels. The last consonant usually was very indistinctly called and is, therefore, treated separately. As the table shows, articulation with noise was distinctly worse than without and articulation with hydrogen and noise was definitely worst.

In order to determine the influence of the hydrogen alone the articulation in hydrogen was divided by the corresponding articulation in air. The result was that the average single-sound-articulation in hydrogen is *ca.* 75 percent of that in air.

There was a small, probably insignificant difference between vowels and consonants and a slightly more marked difference between the relative articulations with and without noise in the speaker's ears. The latter could perhaps indicate, that the speaker tried to correct his pronunciation in hydrogen, when he was able to hear his own voice. Another explanation would be, that an additional amount of distortion, in this case bad pronunciation, has a larger effect at 70 percent articulation than at 90 percent.

It has not been attempted to determine the actual concentration of hydrogen in the cavities. Surely it was not 100 percent because it is hardly possible to speak with 0 percent oxygen in the lungs. With the technique used, the hydrogen was diluted by the residual air so the average concentration was perhaps 80 percent H_2. But if a mixture of 80 percent H_2 with 20 percent O_2 was breathed and the lungs were thoroughly equilibrized with it, the alteration of the voice was less than with hydrogen alone following the above mentioned technique. Thus it seems that the H_2 concentration was more than 80 percent in the gas actually used for speech and consequently the elevation of the resonance frequencies more than one octave.

The figure 75 percent articulation is the result of several factors affecting the articulation. First of all, it is clear that a certain transposition of

[9] Trendelenburg and Wullstein, Preuss. Akad. d. Wissensch., Sitzgsber., Phys.-Math. Kl. (1935), p. 399.

speech sounds is possible and that the frequency level of the whole formant region may be changed without affecting the sound very much. But it seems reasonable that we more easily understand speech sounds of a more usual frequency level because we are accustomed to it. A further factor diminishing the articulation is the decreasing sensitivity of the ear at higher frequencies, which becomes more important when the formants are transposed 1–2 octaves upwards. The above-mentioned alterations of the radiating properties of the mouth and vocal cords and the changed decrement of the cavities, together with the possible frequency distortion by resonance phenomena in the hydrogen cloud and at the boundary between hydrogen and air ought to diminish the articulation too. Finally, it is not the same thing to breathe hydrogen and air. The caller finished speaking when he felt physical signs of anoxaemia. One single breath of fresh air usually was enough to restore well being. But it is well known that physical signs of anoxaemia occur much later than psychical and psycho-physical signs. Thus it is very likely that the pronunciation by the caller was worse in hydrogen than in air. To test the influence of this point it would for instance be possible to make articulation tests in pure nitrogen, where no change of the acoustic properties of the voice producing apparatus would occur.

Summarizing, it is possible to say that by speaking with a light gas a transposition of the speech sounds is effected with a much smaller decrease in articulation than would be expected from former experiments, especially in the case of consonants.

The author is very well aware that the articulation test described is insufficient and that it is necessary to get reliable information on how much the pitch of the formants is really changed. This could be done by means of a high grade oscillograph or some sort of frequency analyzer. The articulation tests should be carried further by a trained testing crew and extended to the single speech sounds. The author has no facilities for doing this.

Detection and Location of Laminations in Steel Plates

HARVEY C. HAYES
Naval Research Laboratory, Anacostia Station, Washington, D. C.
(Received January 5, 1937)

Laminations in the plating and other members of fabricated steel structures cause weaknesses and failures in present welded junctions that did not occur in the case of riveted joints. Thus the need has arisen for some effective method and means for detecting such flaws in structural steel material. Simple theoretical considerations predict that the presence and location of such laminations can be determined by the distortion which they produce in sand patterns formed on the plates when they are thrown into various types of mechanical resonance. Practical tests have substantiated these predictions in the case of plates of different form and thickness.

INTRODUCTION

LAMINATIONS in steel plates do not greatly weaken riveted structures where angle irons and reinforcing strips give added strength to all joints. However, in case of modern welded construction, where such reinforcements are not employed, these defects may introduce weaknesses that are more serious. Thus arises the problem of developing methods and means for detecting laminations in steel plates.

Any method for detecting these lamination defects must be based on measurements of some physical characteristic or property of the plate that is affected by them. The method described herein is based on the effect that laminations produce on the stiffness of the plate, i.e., on its resistance to bending or flecture.

Both the presence and location of the defective or laminated areas are indicated by distortions in a standing flectural wave pattern set up in the plate similarly to the well-known "Chladni's figures." If the plate is vibrated in such a manner that the shape of the sand pattern can be definitely predicted, then the presence of laminations will be indicated by distortion of the sand pattern from the predicted form, i.e., from the form it would take on a perfectly uniform plate.

A lamination distorts the sand pattern through reduction of the stiffness of the plate across the laminated area. Such reduction of stiffness reduces the velocity of the flectural waves traversing the laminated area and this shifts the nodal lines from the position they would have taken on a perfect plate.

THEORETICAL CONSIDERATIONS

The effect produced by a lamination on the stiffness of a plate depends on its depth below the surface. Fig. 1(a) represents a portion of a plate of thickness (a) with a lamination along dotted line $(A-A)$ at a depth (x) from the top surface. Let (S_x), (S_{a-x}), (S_l) and (S_a) represent, respectively, the stiffness of the metal above the lamination, below the lamination, laminated plate and solid plate. Then:

$$S_x = Kx^3, \tag{1}$$

(where (K) is a constant depending on the material, etc.),

$$S_{a-x} = K(a-x)^3, \tag{2}$$

$$S_l = S_x + S_{a-x}, \tag{3}$$

$$S_a = K \cdot a^3. \tag{4}$$

The stiffness (S) of the laminated plate becomes

$$S_l = S_x + S_{a-x} = K[x^3 + (a-x)^3] = K(a^3 - 3a^2x + 3ax^2). \tag{5}$$

Dividing (5) by (4) gives (S_l/S_a), the ratio between the stiffness of the laminated plate and that of the solid plate.

$$S_l/S_a = (a^2 - 3ax + 3x^2)/a^2. \tag{6}$$

Curve 1, Fig. 2, shows the relation between (S_l/S_a) and (x). The minimum stiffness occurs when the lamination is centered between the faces of the plate where it equals one-fourth that of the solid plate. The curve shows that the stiffness of a laminated area remains less than half that of the solid portions of the plate so long as the lamination lies at a depth greater

than about one-fifth of the plate thickness. Under such conditions, the distortion of the sand pattern should be sufficiently definite to indicate the presence and probably the location of the laminated area. A study of the end portions of the plot shows that the method becomes ineffective at an increasing rate as the lamination approaches the surface of the plate and fails entirely when it arrives at or very near to the surface.

Curve 2, Fig. 3, is derived from curve 1 and, as will be seen, applies more directly to the present subject.

The velocity of flectural waves in a plate is proportional to the square root of the stiffness, i.e.,

$$V = c \cdot \sqrt{s}, \tag{7}$$

where (V) is the wave velocity, (s) the stiffness of the plate and (c) a constant depending on the material, etc. The velocity term can be replaced by $(N \cdot \lambda)$ through the well-known relation

$$V = N \cdot \lambda, \tag{8}$$

where (N) and (λ) represent frequency and wave-length, respectively. This gives the relation

$$N \cdot \lambda = c \cdot \sqrt{s}. \tag{9}$$

Since (N), the frequency at which the plate is vibrated, is constant over the whole plate, Eq. (9) states that the wave-length over any part of the plate is proportional to the square root of the stiffness across that area. It follows that the ratio of the wave-length (λ_l) over a laminated portion to (λ_a) the wave-length over a solid portion of the plate is equal to the square root of the ratio of the stiffness over these respective portions.

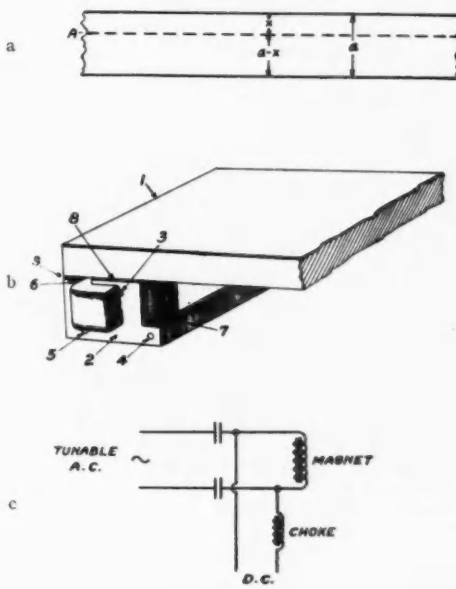


FIG. 1.

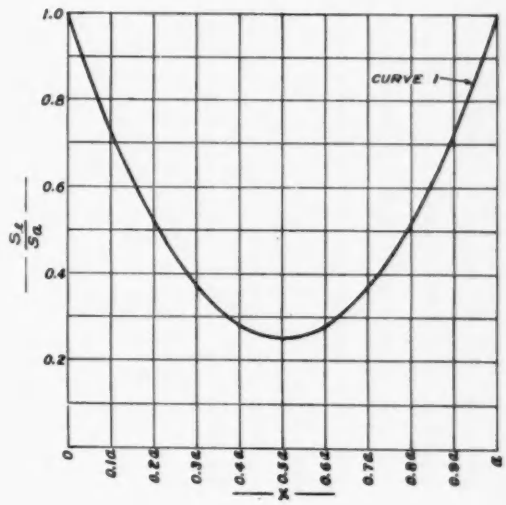


FIG. 2.

The plot on Fig. 3 shows how this ratio (λ_l/λ_a) varies with (x), the depth of the lamination within the plate. If the lamination is located within the center three-fifths of the plate's thickness, this ratio will not exceed one-half and the sand pattern distortion should then be sufficient to definitely show the presence of lamination defects. It is probable that distortions will show for greater values of this ratio, i.e., when the lamination is still nearer to a surface, but they certainly will not be detectable when the lamination is very close to the surface.

It should be noted that plots (I) and (II) are derived on the assumption that the lamination along line $(A–A)$, Fig. 1(a), offers a free shearing plane. Probably slag inclusions adhere to the metal surfaces sufficiently to prevent these curves for representing practical conditions. Such inclusions will not decrease the stiffness of the plate as much as will the free shearing plane assumed and as a result the sensitivity of the method in practice may be expected to be less than that predicted from these curves.

Methods

A plate set into flectural vibrations by bowing, as practiced by Chladni, automatically takes on one of its numerous natural or resonant frequencies. It then vibrates in a standing wave system and sand or powder sprinkled on the horizontal surface will collect along the nodal lines because there is no motion at these locations to scatter it elsewhere. If a plate is forced to vibrate by coupling it to some mechanically driven member, it will take on a standing wave system only when the frequency of the driving force is adjusted to equality with one of its natural or resonant frequencies. Such adjustment of the driving frequency is readily made by noting the point at which sand sprinkled on the plate collects into clean-cut nodal lines.

Portions of the plate on opposite sides of a nodal line oscillate 180 degrees out of phase and as a result the plate is warped back and forth across these lines. Thus it is that the stiffness of the plate is brought into play. If the sand pattern is such that the nodal lines are parallel with one another, the perpendicular distance between two adjacent lines is a half wave-length. It becomes desirable for our purpose to vibrate the plate in a mode that will give parallel nodal lines.

Method Applied to Circular Plates

The nodal lines on a circular plate driven at its center are concentric circles and therefore parallel. Figs. 4(a) and (b) and Fig. 5(a) show three different sand patterns formed on a circular aluminum plate driven at its center point. The driving element was a nickel tube, one end of which was brazed to a lug for screwing it to the center of the plate. This element was surrounded by a magnetizing coil supplied with a.c. current by an electron tube oscillator that could be uniformly tuned through a considerable range of frequency. Each figure represents a definite resonant frequency of the plate. Fig. 4(c) shows the bottom side of this plate. A strip of metal (M) has been cemented to the plate to serve as an artificial defect. Fig. 4(d) shows the distortion it produced on the sand pattern. Here Fig. 4(a) is the predicted pattern—the pattern given by a perfect plate—and Fig. 4(d) is the test pattern. It gives convincing evidence that the plate is imperfect in the southwest quadrant, the quadrant containing the defect.

Figs. 5(b), (c) and (d) all refer to the same plate, a circle 12 inches in diameter cut from a supposedly laminated steel plate. The lack of symmetry in all three figures shows the plate to be defective in the lower left-hand quadrant, while Fig. 5(b) indicates that the defect carries across to the lower right-hand quadrant. This diagnosis is substantiated by examination of the machined edge of the disk. There is definite evidence of a lamination extending from (1) to (2), and some evidence of faults along the arc subtended between (3) and (4) as marked on the margin of Fig. 5(c).

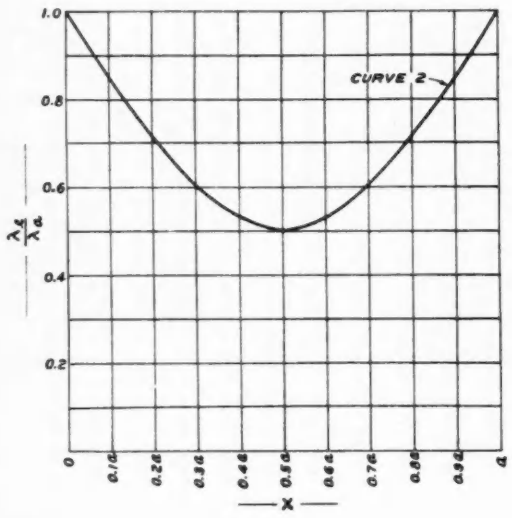


Fig. 3.

Method Applied to Rectangular Plates

The various patterns taken on by a rectangular plate energized at its central point or, indeed, at any single point, are complicated and difficult to predict and in general the nodal lines are not parallel. Distortion of the nodal lines which the presence of a lamination might produce are less readily detectable in such a complicated wave pattern than they are in a simple pattern where the nodal lines are parallel. By driving a rectangular plate along a line extending the whole width, and preferably at the end of the plate, simple grid-shaped patterns can be produced such as are shown in Fig. 6.

The scheme employed to drive the plate is shown in Fig. 1(b), wherein numeral (1) repre-

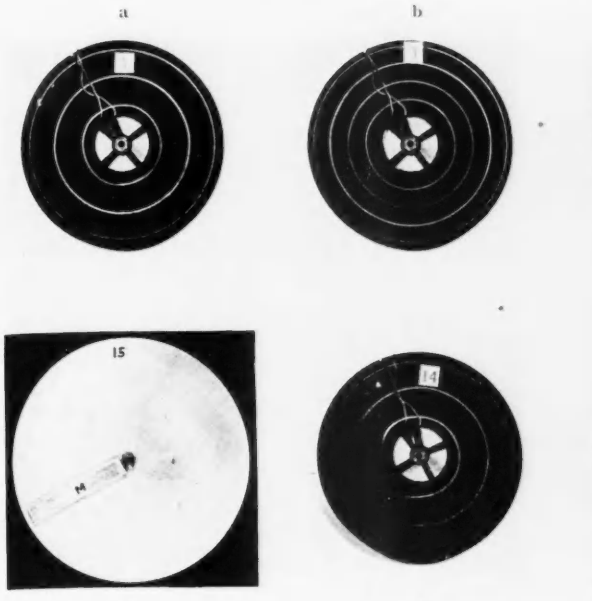


FIG. 4.

FIG. 5. (Lower left and right arrows in *c* should be numbered 2 and 4, respectively.)

sents a rectangular-shaped plate cut away except for the driven end; (2) the core of an electromagnet built up of thin silicon steel punchings piled to a depth slightly greater than the width of the plates to be tested and properly aligned by inserting three small rods through the three punched holes (3, 4 and 5). An energizing coil (in part cut away) surrounds the right-hand or larger portion of the core. It fits into slot (6) on the one side and rests on ledge (7) on the other.

The test plate forms an armature across the jaws of the magnet. A strip of felt (8) separates the plate from one jaw of the magnet, thereby leaving a thin air gap (9) between the plate and the other jaw. The magnetizing coil is supplied with d.c. current through a suitable choke coil for polarizing purposes and a.c. from a tunable power-tube generator. Fig. 1(c), shows the wiring diagram schematically.

In practice, the end of the plate is subjected to bending strains when the coil carries d.c. current because the magnetic flux linkage between plate and magnet core tends to close the intervening gaps. The felt pad (8) takes this strain and therefore acts as a fulcrum with respect to the attractive force across gap (9). The bending strain on the plate across this fulcrum is steady so long as d.c. only traverses the magnet coil, but when a.c. also traverses this coil, the bending strain fluctuates in accordance with the frequency of the a.c. current. When this frequency is adjusted to a resonant frequency of the plate, the sand will form in nodal lines parallel to one another and parallel to the end of the plate, providing the plate is homogeneous.

Theoretically, such a pattern will form for each driving frequency that makes the length of the plate a whole number of half wave-lengths. Therefore, starting at any resonant frequency of the plate, the next one above or below this frequency will, respectively, add or subtract one nodal line. The several parts of Fig. 6 form such a series. Table I gives data referring to these figures.

These patterns were formed on a strip cut from a steel plate 9/16 inches thick. The strip was $5\frac{1}{3}$ feet long and 1 foot wide at the ends, but concave along the free edge, thereby making the width about 11.5 inches at the center. In addition to this lack of true geometric form, the strip was bowed lengthwise and warped or twisted about its longitudinal axis. Moreover, the plate from which it was cut was suspected of carrying laminations. The fact that such regular and consistent sand patterns form on a plate that departs so much from a plane rectangle, as required by theory, argues that there should be no difficulty

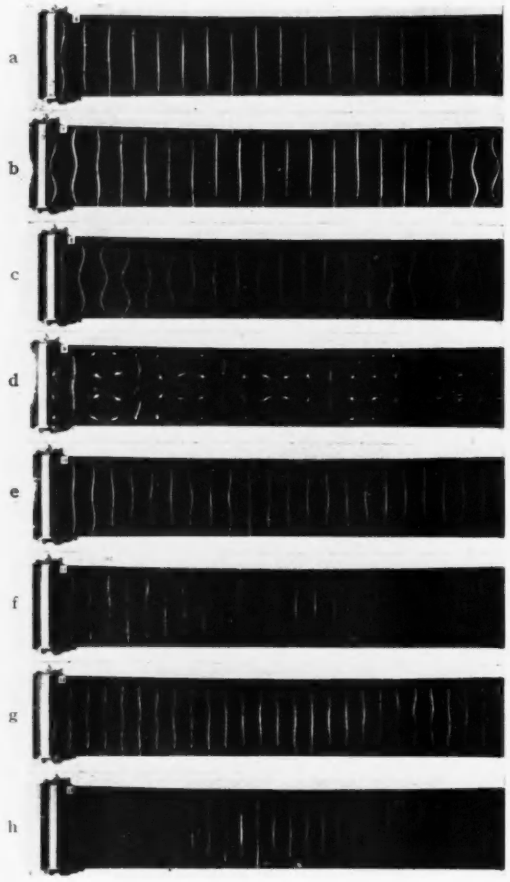


FIG. 6.

in producing such simple grid-shaped patterns on commercial plates.

It will be noted that two patterns are missing from the series, one carrying 23 and the other carrying 26 nodal lines. The reason can be deduced from Fig. 6(d), showing the presence of two wave systems, one carrying 22 nodal lines lengthwise and another carrying 4 nodal lines crosswise on the plate. If the length and width of the plate happen to be such that the half wavelength for some frequency is a common divisor of both dimensions, then the two wave systems can and usually do exist simultaneously. The sand pattern then becomes small dots defining the corners of squares. It might well be termed a "checker-board pattern." Such patterns tend to form when the half-wave is not exactly a common divisor of length and breadth and then show more or less distortion from the pure checker-board pattern. Fig. 6(d) is such a pattern. At frequencies where the half-wave lacks too much from being a common divisor of length and width of the plate to permit both wave systems to exist but does not lack enough to permit either system to exist free of reaction from the other, then neither system can be established. Such conditions doubtless account for the two missing patterns.

As stated, the large plate from which our test strip was cut was suspected of carrying lamination defects. However, it is not known that this particular portion was defective. The sand patterns show no irregularities that would indicate lamination defects. Starting with Fig. 6(a), both end lines show distortion. Fig. 6(b) shows distortion of three lines at each end of the plate and Fig. 6(c) shows distortion of at least seven lines. Finally, the double wave system (where all lines are distorted) appears in Fig. 6(d). The increasing distortion as the frequency is raised to give these successive figures is due to the increasing tendency of the cross-directed wave system to form. And these distortions affect more and more of the lines at each end of the plate as the frequency is raised, because, as stated, the ends are wider than the center. The point to note is that the distortions in each case are symmetrical with respect to a cross line half-way between the ends of the

TABLE I.

PARTS OF FIG. 6	RESONANT FREQUENCIES KILOCYCLES	NUMBER OF NODAL LINES	WAVE-LENGTH INCHES	REMARKS
(a)	4.20	19	7.0	Nodal line at each end distorted.
(b)	4.65	20	6.75	Three nodal lines at each end distorted.
(c)	5.00	21	6.50	Seven nodal lines at each end distorted.
(d)	5.60	22		All nodal lines distorted.
(e)	6.60	24	5.75	Slight nodal distortions on several lines.
(f)	7.20	25	5.50	Slight nodal distortions on several lines.
(g)	8.10	27	5.0	Slight nodal distortions on several lines.
(h)	8.85	28	4.87	Slight nodal distortions on several lines.

figure. A defective plate should not exhibit such symmetry. If this plate carries laminations, either the areas which they separately cover must have dimensions at least as small as the wave-lengths disclosed on the sand patterns or else the laminations do not appreciably affect the stiffness of the plate.

The parts of Fig. 7 refer to a rectangular strip $1' \times 6.5'$ cut from a defective plate $5/32''$ thick. The distorted patterns definitely indicate a large defective area between cross lines (1–1) and (2–2) which is judged to extend across the full width of the plate. Its presence is indicated by the shorter wave-lengths (closeness of the nodal lines) as compared with other portions of the plate area. And the fact that these closer packed nodal lines extend the full width of the plate argues that the lamination covers the whole width. This argument is strengthened by the presence of a lamination showing on each edge over this section. A cut made later across this area proved that the lamination covered the whole width as predicted.

A possible second defective area lies between the cross lines (3–3) and (4–4). Its presence and boundaries are indicated by lack of nodal lines on Fig. 7(e) and distortion of the nodal lines surrounding this area. The patterns, particularly those of Figs. 7(e) and (f), indicate that the defect does not extend to the edge of the plate on either side. Examination of the edges opposite this area showed no laminations.

So far as determining whether or not the plate is defective, any one of the several figures indicates that it is defective because none of the patterns are symmetrical about a cross line midway between the ends of the plate. They have no center or line of symmetry.

Determination of the location and extent of the defect is less definite. In most of the figures the nodal lines are bent and distorted over the whole plate area. These figures differ so much from the simple grid-shaped patterns of a perfect plate that they cannot be interpreted as to the cause of the distortion. Some figures, however, such as (e), (f) and (g), retain sufficient of the grid-pattern to indicate more or less accurately the location and extent of the defect.

Such interpretable figures are given for any frequency that will give a standing wave system for both the plate as a whole and for the defective

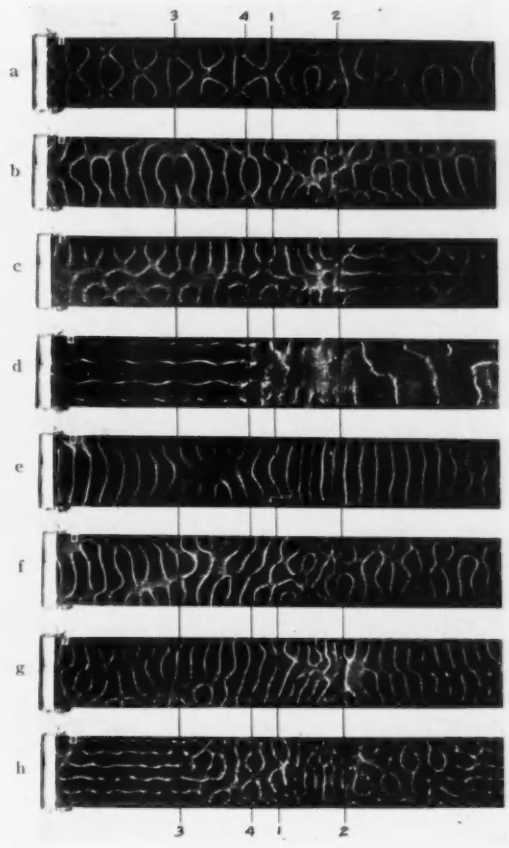


FIG. 7.

area as a whole. The grid pattern then tends to form over both the perfect and defective areas of the plate and the location and area of the defect is indicated by crowding of the nodal lines over the defective area and distortion of those adjacent to this area. Since such figures require a frequency that resonates both the whole plate and the defective areas, they will be noticeably less numerous than is the case for a perfect plate. Indeed, it may sometimes prove that no such figures will be formed on a plate carrying several defective areas. In such a case, the plate can definitely be judged as defective through lack of symmetry of the patterns but the location and extent of the defects will not be clearly indicated.

CONCLUSIONS

Theoretical considerations lead to the belief that:

(1) Laminations in steel plates can be detected

through the resulting reduction of the stiffness of the plate over the defective areas.

(2) The relative stiffness across all increments of the plate area can be deduced from sand patterns formed when the plate is vibrated at one or more of its resonant frequencies.

(3) The presence, location and extent of lamination defects can be predicted with most assurance if the plate is vibrated in a way to give simple sand patterns wherein the nodal lines are parallel.

These predictions have been proved in practice. Circular plates driven to oscillate from the center point give circular parallel nodal lines. Such geometrically perfect patterns on a perfect plate are markedly distorted by an artificial defect such as pasting a small strip of metal on the back of the plate. Another circular plate known to carry a lamination defect clearly showed distortion of the nodal lines from the true circles that would form on a perfect plate.

Rectangular plates one foot wide and of various lengths and thicknesses have been resonated to give simple grid-shaped patterns wherein the nodal lines are parallel and directed crosswise of the plate. One plate known to carry lamination defects gave highly distorted patterns and another plate suspected of carrying defects gave wholly normal patterns.

Tests on these and other plates seem to prove that:

(1) A perfect plate always gives a sand pattern that is symmetrical with respect to a cross line that bisects the plate.

(2) Lack of such symmetry on any clean-cut pattern proves the plate to be defective, though it may not show the location and extent of the defect.

(3) Usually some one or more of the numerous possible patterns will serve to locate the defect and roughly determine its contour.

There is no obvious reason why the method cannot be applied to plates of any size. The nature of the magnet is such that it can be made to cover the width of any plate. Its flexibility allows it to fit the cross contour even though the plate is considerably warped. The attraction between plate and magnet is sufficient to warp the magnet to fit the plate and thus leave a proper and uniform air gap over its whole length.

Tests have shown that 500 watts of a.c. is enough power to oscillate a plate one foot wide and of any desired length or thickness. A 500-watt driver would serve for testing most plates. Two 500-watt power tubes in push-pull relation for amplifying the output of a master oscillator would make an ideal power source. It should be designed to tune through the frequency range 2–10 kilocycles. The magnet, of course, should be designed to match the impedance of the driver. The design as a whole represents a straightforward engineering problem that can be solved by a competent radio engineer.

The Vibrating String Considered as an Electrical Transmission Line

WINSTON E. KOCK*
The Indian Institute of Science, Bangalore, India
(Received August 12, 1936)

An analogy is drawn between the piano string and an electrical transmission line, utilizing a different correlation than is customary in usual electroacoustical analogs. The current in the line is compared with the displacement and the voltage is compared with the momentum. With this correspondence the highly developed treatment of transmission lines becomes applicable to the piano string. The various cases discussed include: Open circuited line with and without attenuation (i.e., ideal and actual string with ends rigidly fixed), terminated line (string affixed to bridge of soundboard), electrical impulse introduced (impact of piano hammer). A relation is derived giving the optimum position of striking point for a given ratio of hammer mass to string mass, and this is shown to agree with the results of George and Beckett. The appearance of the nth harmonic when the string is struck at $1/n$th of its length is explained, likewise the advantages of the modern practice of high string tensions. Conclusions are drawn concerning impedance matching (string to soundboard) and the velocity of propagation for different frequencies.

1. INTRODUCTION

THE theory of the action of the piano hammer and string was first set forth by Helmholtz and extended by Kaufman,[1] Raman and Banerji,[2] Das[3] and others.[4] In attempting to take into account as many factors as possible, the more recent extensions have become rather involved, and it was decided to attack the problem from a somewhat different angle. By making certain changes in the correspondences usually found in electroacoustical analysis, it has been possible to consider the piano string as equivalent to an electrical transmission line. Such an analogy has been known for some time[5] but it has not been developed to the fullest extent, possibly due to the required change in correlations mentioned above. In this paper the details of the analogy are set forth and various conclusions are drawn which are interpretable either in terms of the transmission line or vibrating string. The notations used are listed in Table I.

TABLE I.

Transmission Line	*Piano String*
i = Current	y = Displacement
E = Voltage at a point on the line.	p = Momentum of unit length of string.
L = Inductance per unit length.	ρ = Mass per unit length.
C = Capacity per unit length.	C = Compliance per unit length = $1/T$. (T = Total tension on the string.)
L_h = Equivalent inductance of hammer.	m = Effective mass of hammer.
$L_t = lL$ = Total inductance of line.	$M = \rho l$ = Total mass of string.
l = Length of line or string.	a = Distance of striking point from near bridge.
v_0 = Velocity of propagation.	λ = Wave-length = v_0/f.
$\omega = 2\pi f$ = Periodicity.	

2. DERIVATION OF THE WAVE EQUATION, IDEAL STRING

Consider a string stretched between two rigid supports and assume that there are no frictional or dissipative forces. A wave front W in Fig. 1 is propagated along the string S in the direction of increasing x as shown by the arrow.

At x the displacement is y, at $x+dx$ it is

$$y-(\partial y/\partial x)dx.$$

The difference in displacement along dx is therefore $-(\partial y/\partial x)dx$ and is proportional to the displacing force, mass x acceleration $=\partial(mv)/\partial t$. The factor of proportionality is called the compliance C and as it is per unit length, for the length dx it is Cdx. For a thin string, considered perfectly flexible, this is simply the reciprocal of the tension dx/T and we therefore have

$$-\frac{\partial y}{\partial x}dx=(Cdx)\frac{\partial(mv)}{\partial t}$$

* Of the Research Staff of the Baldwin Piano Company, Cincinnati, Ohio.

[1] W. Kaufmann, Ann. d. Physik **54**, 675 (1895).

[2] C. V. Raman and B. Banerji, Proc. Roy. Soc. **A97**, 99 (1920).

[3] P. Das, Proc. Ind. Ass. Science **1**, 13 (1921).

[4] For references see *Handbuch der Physik*, Bd. 8 Akustik, chap. 5, pp. 177 ff.

[5] For example, Pupin utilized the correspondence between a string with equally spaced loads and a telephone cable with inductances introduced at equal distances in his theory of loading in communication lines.

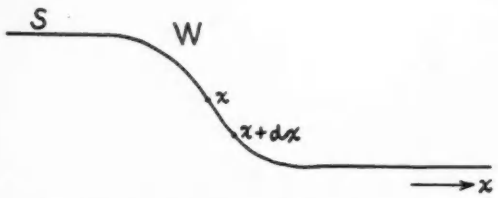


FIG. 1. Propagation of a wave front along the string.

$$\text{or} \qquad -\frac{\partial y}{\partial x}=\frac{1}{T}\frac{\partial p}{\partial t}. \qquad (1)$$

The momentum of the point x is p and at $x+dx$ it is

$$p-(\partial p/\partial x)dx,$$

so that for the element dx, the momentum is $-(\partial p/\partial x)dx$. This is equal to the mass of the element times its velocity of displacement, that is

$$-\frac{\partial p}{\partial x}dx=(\rho dx)\frac{\partial y}{\partial t}$$

$$\text{or} \qquad -\partial p/\partial x=\rho\partial y/\partial t. \qquad (2)$$

Differentiating (1) with respect to x and (2) with respect to t and eliminating p, we have

$$\partial^2 y/\partial x^2=(\rho/T)(\partial^2 y/\partial t^2), \qquad v_0=(T/\rho)^{\frac{1}{2}}. \quad (3)$$

If in the above derivation the electrical terms are substituted for their mechanical equivalents, the usual method of deriving the wave equation for the transmission line is obtained;[6] the equation for the current is then

$$\partial^2 i/\partial x^2=LC(\partial^2 i/\partial t^2), \qquad v_0=(LC)^{-\frac{1}{2}}. \quad (4)$$

It is now seen why the customary electro-acoustical correspondence displacement $\leftrightarrow$ charge and velocity $\leftrightarrow$ current are not applicable here. Eq. (3) is the equation of a string with ends rigidly fixed, therefore at $x=0$ and $x=l$ the displacement is zero. Similarly in an open circuited transmission line at $x=0$ and $x=l$ the current is zero, so that for the exact analog, displacement and current must be correlated. To arrive at the usual correspondence, the mechanical variables displacement and momentum in this case must be differentiated with respect to time. Then displacement $\leftrightarrow$ current becomes velocity $\leftrightarrow$ current and momentum $\leftrightarrow$ voltage becomes force $((\partial/\partial t)(mv)) \leftrightarrow$ voltage.

3. Actual String with Ends Rigidly Fixed

This case corresponds to an open circuited line with attenuation and is best solved by assuming the current and voltage to be simply periodic functions of time[7] $i=Ie^{j\omega t}$ and $e=Ee^{j(\omega t+\theta)}$, where $j=\sqrt{-1}$ and θ is the phase angle difference.

Since any impulse (such as the hammer impact) can be analyzed into a series of pendular Fourier components, no loss in generality is suffered by this process. With attenuation present, the impedance Z and admittance Y will contain, in addition to the reactance introduced by L and the capacitance introduced by C, frictional (resistance) terms, and Eq. (4) becomes $\partial^2 I/\partial x^2=\gamma^2 I$, where $\gamma=(ZY)^{\frac{1}{2}}$ is the propagation constant, and Z, Y, I and γ will in general be complex.

The real and imaginary parts of $\gamma=\alpha+j\beta$ are called the attenuation constant α and the wave-length constant β. The first is so called because it is a measure of the decay of energy (i.e., attentuation) as the wave moves along the line, the second because

$$2\pi/\beta=\lambda, \text{ the wave-length.}$$

Ray[8] has discussed the case of the velocity of propagation for different frequencies on a string with no dissipation. An examination of β indicates that, for the actual string, frictional forces preclude the possibility of equal velocities at all frequencies. For when dissipation is present the velocity of propagation $v_0=\lambda f=\omega/\beta$ does not reduce to $(LC)^{-\frac{1}{2}}$ as in case (1), but remains dependent on ω due to the fact that resistance terms are present in β. The result is that the higher frequencies are propagated with greater velocity and since the wave-lengths for the successive overtones are exact integral multiples of the fundamental (i.e., the modes of vibration of the string are fixed), their frequencies are anharmonic to the fundamental.[9] This is undesirable as it impairs the tone quality.

[6] See, for example, Fleming, *The Propagation of Electric Currents*, 4th edition (Constable, 1927).

[7] For a treatment of the transmission line, see Fleming, reference 6 and Everitt, *Communication Engineering* (McGraw-Hill, 1932), Chaps. 5 and 6.

[8] S. Ray, Phys. Rev. **28**, 229 (1926).

[9] Observed by W. Lange, Hochfreq. Technik u. Elekt. Akustik **45**, 165 (1935).

This condition is also objectionable in electrical communication,[10] and is overcome by the process of loading or by a change over from underground cable to overhead lines. The latter procedure results in an increased separation between wires which increases the inductance and decreases the capacity. This is equivalent to increasing the string tension and mass per unit length of the string and is one of the reasons for the more pleasing quality of modern pianos, made possible by the introduction of massive iron frames to permit greater string tensions and masses.

The quantity $Z_0=(Z/Y)^{\frac{1}{2}}$ is called the characteristic impedance and is that impedance with which a line must be terminated to prevent reflection, that is, it is the termination impedance giving maximum energy transfer. This quantity is seen to depend upon the constants of the line and in order to transmit the same power at lower current, it is necessary to increase this characteristic impedance, for example by decreasing the capacity. In terms of the vibrating string, this means that an increased string tension will permit the same energy transmission with lower amplitudes of displacement. Lower displacement means less deviation from linearity (anharmonic overtones) and less energy dissipated in frictional forces (corresponding to copper loss being proportional to current *squared*) and indicates another reason for the practice of high string tensions.

The input impedance of a line to a given frequency depends upon the length of line; for an open circuited dissipationless line this dependence is given by the relation

$$Z_l=Z_0\frac{\cosh j\beta l}{\sinh j\beta l}=-j\left(\frac{L}{C}\right)^{\frac{1}{2}}\cot\frac{2\pi l}{\lambda}, \qquad (5)$$

where l is the length of the line and the j indicates that the impedance is a pure reactance. This relation is shown graphically in Fig. 2; the negative and positive values signify capacitative and inductive reactances respectively.

For the fundamental frequency of a piano string, l is $\lambda/2$ and it is observed from Fig. 2 that in this case the impedance at the center of the string is zero ($l=\lambda/4$), that is, a voltage (hammer blow) applied at that point would produce the maximum current (displacement).[11] As the position of the striking point is moved away from the center, the impedance increases so that the displacement becomes less for a given voltage and finally reaches zero at the rigid ends. For the first partial, the length of the string l equals λ, so that for this frequency the center of the string presents infinite impedance and in this ideal case a hammer blow at that point would produce no vibration of that frequency.

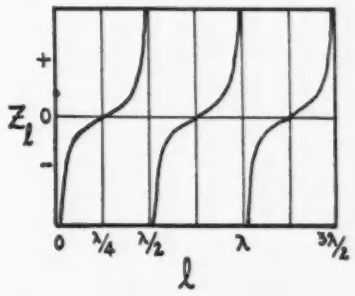


FIG. 2. Variation of impedance with length of line.

In the actual piano string, however, attenuation is present, and the formula for the magnitude of the impedance becomes

$$Z_l=Z_0\left(\frac{\sinh^2\alpha l+\cos^2\beta l}{\sinh^2\alpha l+\sin^2\beta l}\right)^{\frac{1}{2}}.$$

It is seen that only at $l=0$, i.e., at the ends of the string, is Z_l infinite, for as l increases, $\sinh\alpha l$ continually becomes larger so that although $\sin^2\beta l$ may continue to become zero at the nodes, the total denominator is always different from zero. Thus Z_l is always finite and the actual string will accept energy of a given frequency even at the nodes of that frequency.

In practice, the hammer is placed at about 1/7th to 1/9th the length of the string for reasons which will later be observed. For an ideal string this would result in a suppression of the 7th, 8th or 9th harmonic, and Helmholtz suggested this as the reason for such a position. However, these harmonics are not objectionable, in fact they are definitely present in all good piano tones. One result of placing the striking point nearer the

[10] For example, if the velocity of propagation is dependent on frequency, the high harmonics of a square wave telegraphic signal arrive at the receiving end first and cause the pulse to drag out. This necessitates a slower sending rate and delays the transmission procedure.

[11] In part (5), where the effect of the hammer blow is treated more rigorously, it will be observed that other factors are instrumental here.

FIG. 3. Equivalent terminating impedance of a piano string.

fixed bridge is the augmentation of such overtones. This is because the free bridge corresponds to a dissipative effect and, for the wave reflected from this far bridge (the important wave as far as energy is concerned), the attenuation α and likewise $\sinh^2 \alpha l$ would be larger than would be the case were it reflected from the rigid bridge. The impedance of the string to these partials is therefore less and their amplitude accordingly greater.

4. STRING ATTACHED TO BRIDGE OF SOUNDBOARD

This case is the analog of a transmission line terminated with an impedance. For maximum transfer of energy, this terminating impedance should be equal to the characteristic impedance. For a vibrating string with negligible damping,

$$Z_0=(L/C-\omega^2L^2/4)^{\frac{1}{2}}.$$

When the second term is the larger, Z_0 is imaginary; however, the high values of string tension render this term negligible and Z_0 is therefore always real, i.e., $Z_0=(L/C)^{\frac{1}{2}}$, which means that the terminating impedance should be a pure resistance.

For an acoustic radiatior such as the soundboard of a piano, the accession to inertia, which is the additional inertial mass due to the reciprocating flow of air in its neighborhood,[12] can be neglected, due to the fact that the soundboard does not vibrate as a complete unit, but acts somewhat as an acoustic baffle to its various vibrational modes. The reactive component therefore disappears and the acoustic impedance of the air reduces to a pure resistance. The remaining components of the load impedance on the string comprise the compliance, mass, and internal friction of the bridge and the soundboard. The termination is represented in Fig. 3, where C, L and R_i are the bridge and soundboard components and R_r is the radiation resistance.

The bridge is made of very hard wood to minimize frictional losses in it, and due to the heavy string tension, it presents a small compliance. The mass L is kept low by using light wood ribbed for strength; this has the effect of increasing the damping coefficient $R/2L$ and thereby reducing resonance effects and their accompanying undesirable tone quality caused by the presence of formant regions. Likewise R_r is made high by increasing the size of the soundboard; this also increases the efficiency by making the useful component of the load resistance large in comparison with the dissipative frictional effects in the bridge and soundboard.

It is seen that the condenser C discriminates against the low frequencies and this accounts for the low intensities of the fundamental in the bass section of the piano. For compactness in upright and small grand pianos, the bridges for the notes of the middle and bass registers are affixed to the soundboard in such a manner that the free vibrating area for the low tones is less than that of a concert grand. Furthermore the end sections of the deep bass strings between the bridge and hitch pins are shortened; this simultaneously shortens the distance between the bridge and the soundboard edge (where it is rigidly supported). All three of these factors reduce the freedom of vibration of that portion of the soundboard which is effective for the radiation of these low frequencies, so that the compliance C is smaller and the low frequency response poorer. This is one of the reasons for the excellent bass quality of a concert grand piano, where C is made as large as is practically feasible. The so-called "floating bridge"[13] is simply a device for applying the pressure of the bridge more towards the center of the soundboard in order to increase the compliance and improve the bass response.

The combination C, L and R in Fig. 3 represents an impedance possessing a broad resonance region, and the position of this region and the sharpness of resonance will vary for different notes of the scale according as the compliance, effective mass and radiation resistance of the soundboard varies for these different notes. For the treble register the effective radiative area will be less but the effective mass will not decrease in

[12] N. W. McLachlan, *Loud Speakers* (Oxford, 1934), p. 54.

[13] Glazebrook, *Dictionary of Applied Physics* (Macmillan, 1923), p. 467.

equal proportion due to the presence of the nearly constant mass of the bridge. Now the radiation resistance of a piston at these treble frequencies is approximately[14] $r=\rho_0 cA$, where ρ_0 is the normal density of the air, C is the velocity of sound in air, and A is the area. Thus for decreased A the resistance will decrease and the decrement $\delta=R/2fL$ (f=frequency) of the circuit of Fig. 3 will decrease with frequency. Inasmuch as a filter circuit becomes more effective as the decrement decreases, the high notes of a piano should be lacking in overtones due to this filtering action, and such is found to be the case.

5. Analysis of the Hammer Impact

The imparting of energy to the string by means of a hammer impact is analogous to connecting across a transmission line an inductance possessing an e.m.f. between its terminals. The momentum of the hammer mv corresponds to the voltage on the inductance $L_h(di/dt)$. For a resilient hammer a condenser would be in series with the inductance and a nonelastic compression of the felt on impact would be equivalent to a leak across the condenser.

As an example of the simplicity in treatment which can be introduced by means of the electrical analogy, an analysis will be given of the optimum position of impact for a given hammer mass. It will be observed that by means of a straightforward procedure, this problem, for which no satisfactory solution has previously been found, can quickly be solved to yield an equation which agrees quite well with experimental observations.

For simplicity we assume an ideal string struck by a hard hammer. The equivalent circuit is shown in Fig. 4. At the instant of impact, switch S is closed and the voltage $L_h(di/dt)$ is impressed across the line. Current waves travel in both directions and are reflected from the ends with a reversal in phase. The longer the inductance is connected to the circuit, the more energy it can impart to the line, up to the point when the reflections from the ends reach it. Since the reflected waves have reversed their phase, they will tend to restore energy to the inductance;

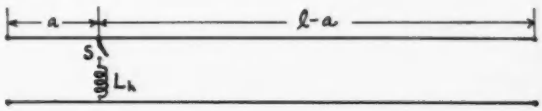


FIG. 4. Equivalent circuit for representing hammer impact.

therefore if the switch is opened just before this occurs, maximum transfer of energy is obtained. The reflected wave from the short end reaches the inductance very soon in comparison with the time of reflection from the far end, and the time of contact would be quite short if the inductance were then removed, and the energy transfer small. The desired effect is therefore obtained by allowing the reflections from the short end to reverse the voltage on the inductance (reverse the direction of motion of hammer) and to have switch S open (to have the hammer quit the string) just before the reflected wave from the far end reaches it. The optimum time of contact is thus equal to twice the length l-a divided by the velocity of propagation. That is

$$[t_c]_{\text{opt.}}=2(l-a)(LC)^{\frac{1}{2}}. \tag{6}$$

For a given initial velocity, the time of contact of the hammer depends upon its mass, the mass of the string, and the position of impact; the heavier the hammer, the longer the duration of impact, and the further the point of contact is moved towards the near bridge, the greater the effect of the tension of the string and the shorter the contact time. These relations are implicitly contained in the expression for the impedance of the open circuited line,

$$Z_l=-jZ_0\cot(2\pi l/\lambda). \tag{5a}$$

From the above consideration the hammer will have left the string by the time the wave reflected from the far end reaches it, therefore the impedance which is effective in this case is that of the short end of the line,

$$Z_a=-j(L/C)^{\frac{1}{2}}\cot(\pi a/l). \qquad (\lambda=2l.)$$

Since, for $a<(l/2)$, $\cot(\pi a/l)$ is positive, the $-j$ indicates that the impedance of the short end will always be a pure condensive reactance, and the equivalent condenser C_a is obtained from

$$Z_a=1/j\omega C_a=-j(L/C)^{\frac{1}{2}}\cot(\pi a/l).$$

[14] I. B. Crandall, *Vibrating Systems and Sound* (Macmillan, 1927), p. 149.

Recalling that

$$\omega = 2\pi f = 2\pi v_0/\lambda = 2\pi(1/LC)^{\frac{1}{2}}/2l = \pi/l(LC)^{\frac{1}{2}},$$

we obtain $$C_a = (lC/\pi)\tan(\pi a/l). \qquad (7)$$

We can thus replace the picture of a transmission line with its reflections by its equivalent, an inductance and a condenser connected in parallel. Then instead of many reflections there will be one surge of current which will reach a maximum and return to zero, and, if the inductance remained connected, the energy would continue to oscillate back and forth between inductance and condenser. Likewise the voltage $L_h(di/dt)$, a maximum at the instant the switch is closed, goes to zero as the hammer reverses its direction and becomes increasingly negative up to the point where the hammer leaves the string. The time of contact is thus the time of one-half cycle of the oscillatory circuit composed of L_h and C_a. The natural frequency of this circuit is $f = 1/2\pi(L_hC_a)^{\frac{1}{2}}$, so that one-half the period is

$$t_c = \pi(L_hC_a)^{\frac{1}{2}}. \qquad (8)$$

The relation giving the position of the hammer to produce maximum energy transfer is obtained by equating t_c to its optimum value as given in Eq. (6). We then have

$$2(l-a)(LC)^{\frac{1}{2}} = \pi(L_hC_a)^{\frac{1}{2}},$$

which with (7) gives

$$2(l-a)(LC)^{\frac{1}{2}} = \pi\left(\frac{L_hlC}{\pi}\tan\frac{\pi a}{l}\right)^{\frac{1}{2}}.$$

This can finally be written

$$\frac{4}{\pi}\left(1-\frac{a}{l}\right)^2\cot\frac{\pi a}{l} = \frac{L_h}{L_l} = \frac{m}{M}, \qquad (9)$$

where $L_l = lL = M$ is the total mass of the string. This relation between m/M and a/l can be interpreted either as the optimum position of impact for a given ratio of hammer mass to string mass, or as the optimum hammer mass for a given striking point.

In Fig. 5 is plotted the curve of m/M *versus* a/l for maximum energy transfer. The smooth line is the theoretical curve and the circles are the points as found experimentally by George and Beckett for various values of m/M;[15] since they employed in their experiments hard hammers and rigid bridges (approximating an ideal string), the conditions closely correspond to the case here analyzed and it is observed that the agreement is quite good.

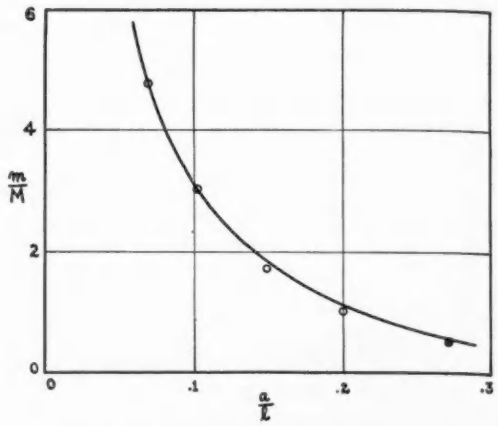


FIG. 5. Curve showing the relation between m/M and a/l for maximum energy transfer. The heavy line is the predicted curve and the circles are the experimental results of George and Beckett.

From the curve it follows that, for values of m/M which are used in practice, the striking point should roughly fall between one-seventh and one-ninth the length of the string, and this is actually the case.

It is interesting to note that, for small values of a/l, formula (9) reduces to the approximate one obtainable by considering the hammer as an additional weight attached to the string at a distance a from the near bridge. Then the equation of motion is approximately

$$md^2y/dt^2 = -Ty/a, \qquad (y = \text{displacement}),$$

so that the frequency of oscillation is

$$f = (1/2\pi)(T/ma)^{\frac{1}{2}},$$

and one-half the period is

$$t_c = \pi(ma/T)^{\frac{1}{2}}.$$

Setting this equal to the optimum time of contact,

$$[t_c]_{\text{opt.}} = 2(l-a)(\rho/T)^{\frac{1}{2}} = \pi(ma/T)^{\frac{1}{2}},$$

[15] W. H. George and H. E. Beckett, Proc. Roy. Soc. **A114**, 111 (1927).

we arrive at the expression

$$\frac{m}{M}=\frac{(4/\pi)(1-(a/l))^2}{(\pi a/l)}. \qquad (10)$$

This corresponds to Eq. (9) for small values of a/l, since in that case

$$\cot\frac{\pi a}{l}=\frac{1}{\tan(\pi a/l)}\approx\frac{1}{(\pi a/l)}.$$

For large values of a/l, however, formula (10) breaks down, due to the fact that in its derivation the effect of the mass of the short end of the string was not taken into account. In the more rigorous derivation, this factor entered into expression (5) from which the equivalent total capacity C_a (corresponding to a/T above) could then be more accurately obtained.

6. Summary of Results

The conclusions to be drawn from the foregoing analysis may be summarized as follows:

1. By increasing the string tension and mass per unit length, more uniform velocities of propagation of the various partials are obtained, and the distortion caused by the presence of anharmonic overtones is reduced. The increase of mass, however, is limited, due to the fact that the string eventually vibrates as a bar and again introduces anharmonic distortion. Greater string tensions also reduce attenuation and therefore permit greater resounding power; furthermore they permit the transmission of energy with less amplitude of vibration and thus again reduce nonlinearity and anharmonic distortion.

2. The nth harmonic can be excited by an impact at $1/n$th the length of string whenever attenuation is present in the string.

3. An increase in the compliance of the string termination (bridge and soundboard) for the lower registers results in better bass response.

4. A decrease in the effective mass or compliance of the treble register would tend towards increased brilliance.

5. For any ratio of hammer mass to string mass, the proper point of impact for maximum loudness can be calculated for the ideal case (curve of Fig. 5).

Acknowledgment

I should like to express my appreciation for the guidance of Sir C. V. Raman in this work.

Direct Recording and Reproducing Materials for Disk Recording

A. C. KELLER
Bell Telephone Laboratories, New York City

(Received January 5, 1937)

Recently materials for direct recording and reproducing work have been improved so that they are now suitable for many uses. These materials, as they are available on the market, are classified chemically into five groups and measurements are given of frequency characteristic, surface noise, life, distortion, etc. These data have been taken with both lateral and vertical recording. A short aural demonstration was given of typical recording on materials of each group.

THERE has been a need for some time for a suitable combined recording and record material of the disk type for use with mechanical recording. Materials for this work must be suitable for recording and after recording must be capable of being played back satisfactorily. As recording materials, they should be capable of good quality recording with relatively simple apparatus. As record materials, they must render reproduced sound of good quality and low surface noise with simple apparatus, and must also have adequate life and be capable of being satisfactorily reproduced with a sturdy reproducer. These requirements are, of course, all relative and will depend upon the technical character of the work to be done and upon the standards of excellence required.

Of course, wax cylinders have been, and are being used for similar types of work for which disks of the kind to be discussed in this paper can be used and these are capable of producing good results. Wax disks have also been used but their applications have, in recent years, been largely confined to sound picture studio and other more elaborate uses. Even though these cylinders and disks of wax have done many types of work satisfactorily, there has been a real need for a thin, sturdy, and easily stored disk of a material on which sound can be recorded and reproduced with rugged instruments. This statement seems justified by the growing number of suppliers of a newer type of record, which usually takes the form of a thin disk from 10 to perhaps 60 mils thick of a variety of materials.

This paper will give data on some of the materials of this type as they are now available on the market for general use. As there are many possible combinations of recording instruments, reproducing instruments, stylii, etc., the data will give a rather general idea of what can be expected from the various recording materials. The materials will be referred to as certain types of distinguishing physical or chemical characteristics and not by their trade names. While it might be interesting to mention the disks by name, the more general classification entirely satisfies the needs of this paper.

The disk record materials which have come to our attention for this type of recording may be classified as follows (see Table I):

Group 1. Materials containing cellulose esters. These materials have been mainly plasticized cellulose nitrate. All of the disks which we have tested which were stated to be cellulose acetate or "acetate" by the suppliers, contained a predominant amount of cellulose nitrate.

Group 2. Metal disks such as aluminum, zinc, lead, pewter, etc.

Group 3. Plasticized thermo-setting phenolic resins which are used soft and uncured as recording materials. A baking operation after recording transforms the material into a hard record material.

Group 4. Various types of resinous compounds other than the cellulosic and phenolic compositions mentioned above.

Group 5. Gelatin and gelatin compositions.

TABLE I.

GROUP	MATERIAL
1	Cellulose esters
2	Metal disks (aluminum, zinc, lead, pewter, etc.)
3	Plasticized thermosetting phenolic resins
4	Resins other than those of group 1 or 3
5	Gelatin and gelatin compositions

Except for the metal disks under group 2, some of the materials of all of these groups are or have been available as coatings on metal disks. A typical coated metal disk will consist of a $\frac{1}{32}$ inch aluminum disk with a 5 to 10 mil coating on either side. Some disks are only available as thin sheets of the recording material and others are available also as coatings. A few have also been supplied as coatings on paper and this appears to be a somewhat less expensive method of preparing the disks, but judged by the relatively few that are or have been on the market, it is apparently not as easy to produce these of equivalent quality of reproduction and ease of use.

These direct recording and reproducing disks can be recorded in a variety of ways. Many of them are recorded with a cutting stylus in much the way that waxes are cut in the phonograph industry. The fact that the materials are somewhat harder than wax makes the use of an advance ball unnecessary. You will recall that the usual advance ball used in phonograph work is a carefully ground portion of a jeweled sphere located adjacent to the cutting stylus of the recording instrument and adjustable with respect to it in such a way that a predetermined width of groove may be obtained even on very soft materials such as wax, etc. The recording styli used for direct recording work are of steel, sapphire or diamond. The life of these is a function of the character of the material itself, the cleanliness of the material and the skill of the operators using them. In general, sapphire and diamond styli are more costly but are harder and have a longer life on many materials than the steel styli now available. The jeweled styli are however more brittle and for this reason are subject to breakage when used with materials which contain small particles of foreign matter.

The shavings which occur in the cutting operation are not generally removed by a suction tube as in wax recording. The continuous thread of material which is obtained with properly adjusted recording conditions is handled in a number of ways. It has been collected, for example, by a radial brush resting lightly on the record surface and it has also been collected when recording by permitting the shaving to coil up inside the position of the cutting stylus on the record.

Some of the coated disks are pregrooved, in a molding operation, by the use of suitable matrices. These disks require a somewhat less expensive recording equipment in that the usual lead screw and associated mechanism is not required to feed the recording device across the surface of the disk during recording. Pregrooved disks have mainly been used for so-called home recording and similar uses whereas many of the other disks discussed here have been used for more critical work. All of the materials discussed in this paper are capable of being reproduced with rugged instruments which are guided across the record surface during reproduction by the recorded grooves themselves without the aid of a lead screw. The pregrooved records are recorded as an embossing operation in which a blunt tool is used in contrast to the cutting tool mentioned earlier. The embossing or rubbing tool or stylus in the recording operation embosses a somewhat wider and deeper modulated groove than the original molded unmodulated guiding groove.

The metal disks under group 2 are also available as pregrooved or plain disks. The pregrooved variety of disks is recorded in the same way as that just described for the pregrooved coated disks. The plain metal surface disks, however, are not generally cut as were the coated disks but rather embossed with a blunt stylus in the manner used for pregrooved disks. A surface lubricant is often used during the recording of metal disks.

The gelatin disks under group 5 are generally recorded with a cutting tool in the manner used for the disks of groups 1, 3 and 4. After recording, gelatin disks are frequently treated chemically in order to harden the record surface. These treated surfaces have a somewhat longer life in reproduction.

For reproducing from these direct recording materials a number of stylus shapes and materials have been used. Stylus materials have been steel, sapphire, diamond and various forms of fiber, thorn, etc. In general, the type of needle used will depend upon the design of the reproducing instrument and the number of times you wish to reproduce the record. It is possible to design reproducing instruments of small

enough mechanical impedance so that pressures of the stylus on the record material can be reduced to a value such that any of the stylus materials mentioned are capable of reasonably satisfactory reproduction provided that the stylus has a slightly rounded point, that is, has a tip radius of the order of 2 mils. Instruments which may be satisfactorily used at these moderate record pressures are a refinement over those available for use in the ordinary electrical home phonograph.

Rapid wear of either the stylus or the record or both has been encountered at the higher record pressures with some combinations of record and reproducer stylus materials. This can also be due to the higher coefficient of friction between these material combinations. For this reason, even with a properly contoured needle (or one which has been approximately contoured by playing a few grooves of a common phonograph record which is somewhat abrasive) certain combinations of stylus and record materials will require lower pressures in order not to injure seriously either the stylus or the record in a few playings. For example, the coefficient of friction between a steel stylus and an aluminum record is about 0.3, which compares with the unusually low value of somewhat less than 0.02 for a fiber stylus and an aluminum record.

Apart from difficulties arising from the coefficient of friction between materials, the contour of the stylus is very important. For example, a new steel phonograph needle is relatively sharp and the area of contact between it and the record is of the order of a few millionths of a square inch, so that with a reproducer resting on a record, the needle point pressures are of the order of 20–100 tons per square inch. Pressures of this order will rapidly wear either the stylus, the record, or most probably both. An abrasive material is included in commercial phonograph records in order to grind the needle to fit the groove of the record and to prolong the record life. In a matter of seconds the needle point pressures of a steel needle playing a common phonograph record are reduced to values of the order of 5 tons per square inch. It is interesting to note that even these lower pressures are very large and somewhat higher than pressures which are known to be the upper limits for which lubricants will remain between surfaces. This is the reason that lubricating phonograph records by various substances has rarely shown any effect on either surface noise or record life. Only for instruments which are capable of being satisfactorily used at very low record pressures can the effect of lubrication be measured.

The nature of the reproducer stylus material and the stylus design will also affect the quality of reproduction even if used at needle point pressures where the effect on either the record material or the stylus is negligible. For example, a needle with a large effective mass will tend to reduce the high frequency response of the reproducer and a similar effect will be obtained by the use of a thin or relatively flexible needle. While the amount of reduction of high frequency response will depend on the effective mass and stiffness of the stylus, it will also depend upon the relationship of these to the vibratory constants of the reproducing instrument itself and upon the relationship of the mechanical impedance of the complete reproducing instrument to the mechanical impedance of the record material. A reduction of high frequency response is, in general, not desirable because it will tend to make the reproduction less natural, at least for many types of recording. However, a reduction in high frequency response will also have the effect of reducing surface noise and this may be important under some conditions.

To give a complete picture of record materials of the type being discussed and the reproduction from them with all available reproducers would make a subject beyond the scope of this paper. Typical data will rather be shown, taken with some of the materials and instruments which are at present available.

Quantitative measurements taken with both lateral and vertical recorders and reproducers at linear speeds of 14 to 20 inches per second will include:

(a) Response *versus* frequency characteristics,
(b) Surface noise,
(c) Wear (life),
(d) Distortion measurements.

Frequency Characteristics

The frequency response has been measured by recording single frequencies and then meas-

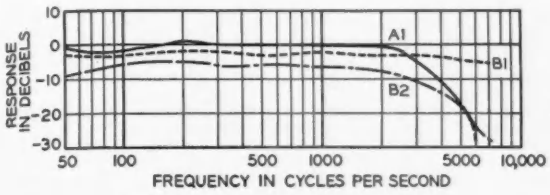


FIG. 1. Lateral recording—group 1.

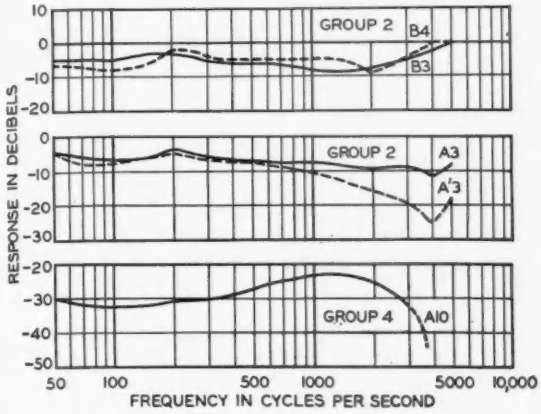


FIG. 2. Lateral embossed recording.

uring the reproducer response as the record is played back. The curves which follow show the *relative* response compared with that obtained from pressings made from processed matrices. The waxes used for producing the comparison matrices were recorded in the same way and with the same recorders used for each of the materials under test. Figs. 1 through 3, inclusive, are frequency response measurements taken with lateral recording. Fig. 4 shows similar measurements with vertical recording.

Fig. 1 shows lateral cut frequency characteristics of materials of the cellulosic type which has been called group 1. The recording has been done with a steel recording stylus. In general, the response differences of this and other groups occur as the frequency is increased. Fig. 1 shows the effect obtained with two reproducers of different design and vibratory characteristics. Curve A1 taken with reproducer *A*, a commercial reproducer fitted with a contoured (about a 2-mil tip radius) steel stylus and material No. 1 shows poorer high frequency response than does curve B1 taken with laboratory reproducer *B* fitted with a contoured diamond stylus on the same record. Material No. 1 is the best of those tested with lateral recording and classified under

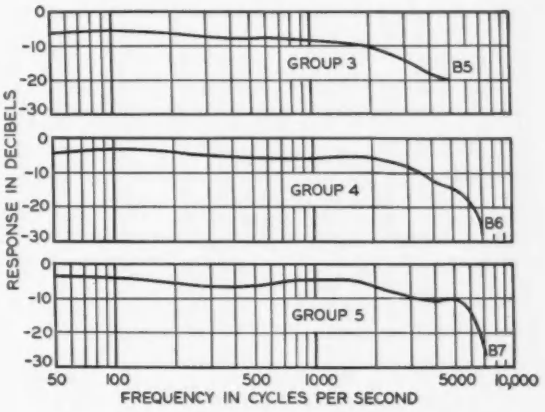


FIG. 3. Lateral recording.

group 1. Fig. 1 also shows the variation of materials of this group as supplied by two manufacturers. Curve B2 has been taken on another material of the cellulosic group which showed the poorest frequency response of the materials tested of this group. You will notice a considerable difference in response in the region of 5000 cycles between these materials and that the better material is down very little in this frequency range compared with the molded record.

Fig. 2 shows lateral frequency response characteristics taken with embossed grooves. These curves are *relative* to measurements taken with molded records made from matrices which in turn have been made from lateral *cut* waxes. Curve A10 has been taken with a pregrooved material of group 4. Prior to recording, the grooves were unmodulated and relatively narrow and shallow. After recording the grooves were similar to those of a standard phonograph record, that is, about $2\frac{1}{2}$ to 3 mils deep and 6 to 7 mils wide. The frequency response of curve A10 shows a relative drop of more than 30 db at 4000 cycles, much poorer than any other material tested.

Curves B3 and B4 of Fig. 2 show the results with an aluminum disk and lateral embossed recordings. This material is probably the most widely used at present of the solid metal group. Curve B3 has been taken with a plain aluminum disk and curve B4 with a pregrooved aluminum disk. In recording, the aluminum surface is often lubricated with a wax such as paraffin, or with kerosene. The use of the lubricant produces a

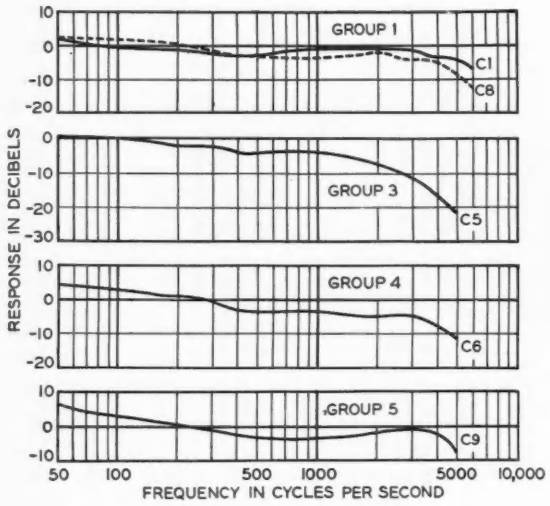


FIG. 4. Vertical recording.

record several db quieter than one which is recorded without a lubricant. These curves were taken with the laboratory reproducer B which is fitted with a diamond stylus and show that with an adequate recording instrument frequencies up to 4000 to 5000 cycles can be satisfactorily recorded on aluminum. The measurements at 5000 cycles and higher were considered doubtful because of the high surface noise levels in this region.

In order to show the effect of the reproducer stylus on the frequency response characteristic curves A3 and A'3 are shown on Fig. 2. Reproducer *A*, although somewhat inferior to reproducer *B* in response and mechanical impedance, lends itself to rapid stylus changes and has been used for this reason. Curve A3 is a remeasurement of the record used for getting curve B3 with a contoured steel stylus in the reproducer. Curve A'3 is another measurement of the same record and reproducer *A* fitted with fiber needle. The difference in response in the region of 5000 cycles can be seen to be considerable. Fiber or thorn needles wear quite rapidly on aluminum records because of their mechanical properties. The characteristic shown is about the best that can be obtained with needles of this kind and instruments now available. As the needle wears, the response decreases at the higher frequencies and the distortion products increase rapidly. Fiber or thorn needles are frequently badly worn before completely playing a single record with a playing time of the order of a few minutes.

Curve B5 of Fig. 3 shows the relative lateral cut recording frequency characteristic of a plasticized thermosetting phenolic material of group 3. The response has been taken after baking the recording for two hours at about 350°F in a small oven. During the baking operation, there is some flow of the material which undoubtedly changes the frequency characteristic of the recording. Curve B5 was taken after baking. Difficulty was encountered at some of the higher frequencies in measuring this material due to high noise levels on some of the records.

Curve B6 of Fig. 3 shows the relative lateral cut frequency response characteristic of a resinous material of group 4. This material is the best of those examined of this group. This group, however, covers a wide range of materials which have very little in common so that Curve B6 must be taken as a measurement of one material only and not necessarily as illustrative of this group.

Curve B7 of Fig. 3 shows the relative lateral cut frequency response characteristic of the best gelatin compound of group 5 that was measured. This group also gave trouble with high noise levels interfering with the measurements at higher frequencies.

Fig. 4 shows the relative frequency response measurements taken with materials of groups 1, 3, 4 and 5 with vertical cut recording. As for the measurements with lateral recording, these data have been plotted to show the *relative* frequency response compared with that obtained from molded records prepared from processed matrices. Here again, the waxes used for producing the matrices were recorded in the same way and with the same recorders used for each of the materials under test. Measurements are not shown for materials of group 2 with vertical recording. Although only a limited amount of work has been done recently with embossing vertically modulated grooves in these materials, their mechanical impedances are generally large enough compared with that of the recorders used to make the measurements of doubtful value as performance ratings. Curves C1 and C8 of this figure taken with vertical reproducer *C*

fitted with a diamond stylus, show the approximate range of frequency characteristic of materials of the cellulosic group (group 1) measured with vertical instruments. Curve C1 has been taken with material No. 1 and is the same material which was used for the lateral recording shown in curves A1 and B1 of Fig. 1. The poorest material of group 1 for vertical recording, however, was not found to be same material as the poorest for lateral recording.

Curve C5 shows the relative vertical cut frequency response characteristic of a material of group 3. The response is down about 20 db at 5000 cycles.

Curve C6 shows similar data for a material of group 4. The relative response is down about 10 db at 5000 cycles.

Curve C9 shows similar data for a material of group 5. The relative response is down about 8 db at 5000 cycles.

In taking these data on frequency characteristics, electrical equalization has not been used, in either recording or reproducing, in order to obtain the over-all frequency response characteristic. It is possible to equalize within reasonable limits at either step and thereby to improve the frequency characteristic. When equalizing in recording, particularly at the higher frequencies, care must be taken to avoid excessive curvatures in the grooves for increasing recording levels because these will give rise to severe distortion. When equalizing in reproducing, care must be taken to avoid excessive correction because this will increase surface noise.

The value of frequency response measurements, although very great, does not give all of the important facts which are necessary in quantitatively rating the performance of recording materials. Two other very important items are surface noise and distortion.

Surface Noise

A tabulation of the data on surface noise for both lateral and vertical recording is shown in Table II. The noise measurements have been taken with a noise meter having a frequency weighting network and a damped meter of the volume indicator type. While this kind of a noise measurement does not precisely check ear measurements, it is sufficiently accurate for the record measurements being taken here and has the advantage of being simple and quick, which is important because of the large number of measurements needed. The values are given in db and are referred to a convenient arbitrary zero. The larger the db value given, the noisier the record.

The measurements are again divided into five groups according to the material groups and show a range of surface noise from the minimum value measured to the maximum measured, not, however, necessarily on materials of a single supplier. The range of values measured on the quietest record in each group is also given.

The lateral measurements have been taken with reproducer *A* equipped with a steel stylus except where noted and with a record pressure of 50 grams. The vertical measurements have been taken with reproducer *C* equipped with a diamond stylus and at a record pressure of about 30 grams. The lateral and vertical noise levels are not exactly comparable because the vertical reproducer used had about twice the frequency range of the lateral reproducer. These noise measurements show some of the records of group 1 as lateral records to be somewhat quieter than a commercial phonograph record. The surface noise level of vertical recording with materials of group 1 measured with a vertical reproducer occasionally approach the level of a quiet molded vertical cut transcription record. These measurements have been taken without equalization in the reproducer circuit to give a uniform frequency response characteristic. Such equalization for the characteristic measured on these materials will increase the measured noise levels.

The measurements listed in Table II except those of group 2 were all taken on records which were cut with the best steel recording styli tested, which closely approached the results obtained with sapphire styli. The measurements on records of group 2 were taken on records which were embossed with a diamond stylus.

The noise levels on a given material with steel recording styli as received from several suppliers showed a variation of more than 10 db.

In many cases, it has been difficult to obtain consistent data. Sometimes this has been due to variations in the material itself and at other

times to the effect of the recording materials on the recording stylus. Some recording materials contained abrasives which caused rapid wear of the recording stylus.

Wear Data

The record life has been measured in concentrically recorded unmodulated grooves with lateral and vertical instruments and is shown in Table III. The record life has been taken as the number of times the groove is played for a 5 db rise in noise level. Measurements were not carried beyond 500 playings. It has been assumed that the recorded levels on the records are such that a 5 db rise in surface noise is reached before the record is judged to be worn out due to the wear on modulated grooves. This is not true for records made with excessive recording levels or with records reproduced with instruments of high mechanical impedance. The reproducers and styli used for these tests were the same as those used in the surface noise tests. A number of measurements were obtained which indicated a record with a life greater than 500 playings. Some of these records, however, started with a high initial noise level and were so rough at the start that they showed little noise increment during test. These are specifically noted in the table. Omitting these records, many showed good wearing qualities adequate for many uses, even though not entirely comparable with commercial molded records. A life greater than 500 playings has been measured for commercial phonograph records. This is due to the low (50 gram) value of record pressure used. For a record pressure more commonly used of about 125 grams, this figure is reduced to 50 to 100 playings.

Distortion Data

It is difficult to give a simple rating of distortion which will satisfy all conditions and which will rate the various materials as a trained observer would for this factor if listening to speech or music. As a simple rough measurement data have been taken of the harmonics found in the reproduced output of a 1000-cycle recorded tone.

The 1000-cycle data are tabulated for both lateral and vertical recording in Table IV. These data show the materials all to be somewhat inferior to molded records. The measurements have been corrected for frequency response characteristic. If equalization for frequency response had been used in recording the distortion

Table II. *Surface noise data in decibels.*

Material Group	Lateral Recording Reproducer *A*, Steel Needle 50 Gram Pressure		Vertical Recording Reproducer *C*, Diamond Stylus 30 Gram Pressure	
	Range	Best Record	Range	Best Record
1	10–45	10–13	11–52	11–13
2	29–38*	29–32*	18–26	18–26
	33–39	33–36		
3	25–32	25–28	25–31	25–27
4	20–33	20–27	24–36	24–30
5	27–38	27–29	28–37	28–30
Lateral Cut Phonograph Record	23–40			
Vertical Cut Electrical Transcription			10–15	

* Measured with a fiber needle.

Note: The lateral and vertical record noise levels are not exactly comparable because of the wider frequency range of the vertical recording.

Table III. *Record life on unmodulated concentric grooves.*

Material Group	Lateral Recording Reproducer *A*, 50 Gram Pressure, Steel Needle	Vertical Recording Reproducer *C*, 30 Gram Pressure, Diamond Stylus
1	5—>500‡	12—>500
2	5—12*	8—25
	2—4	
3	22—>500	26—>500
4	39—211	32—>500
5	4—>500‡	235—305
Lateral Cut Phonograph Record	>500	
Vertical Cut Electrical Transcription		>500

‡ High initial surface noise level.

* Measured with a fiber needle.

Note: Vertical reproducer *C* has approximately twice the frequency range of lateral reproducer *A*.

would be greater than the tabulated measurements. The reproducers used for these measurements were reproducer *A* for the lateral and reproducer *C* for the vertical data.

Power Requirements

All materials tested except aluminum and the pregrooved material of group 4 shown in curve A10 of Fig. 2 required power inputs to the recording instrument at 1000 cycles that were only a few db more than that required for wax recording. Aluminum depending upon the source of supply required about 5 to 15 db more recorder input power at 1000 cycles than did the control wax recording. The group 4 material of Fig. 2 required about a 20 db increase in power input to the recorder.

The increase of turntable power required has been found to be relatively moderate as compared with wax recording for most materials. Here again, aluminum required a considerable increase in turntable power. No exact measurements of this have been made but it is evident that the increase will be a function of the depth of the recorded groove.

Processing

A few of the better records tested have been recorded and processed, that is, matrices have been electroplated from them and molded records made from these matrices. These few have shown an increase in surface noise in the processing of the order of 10 db. Some of the other materials are not suitable for the present type of processing and would probably be re-recorded to wax if matrices are required.

Fire Hazard

The materials of the cellulosic group (group 1) are most frequently supplied as coated metal disks and are usually recorded by a cutting process. The shavings from all of the materials tested of this group were found to be inflammable due to the large amount of cellulose nitrate contained in them. The coated disks themselves are not readily inflammable because it is necessary to raise the metal core and the coating to kindling temperature in order for the record to burn, and this requires relatively long exposures at fairly high temperatures. The shavings from all other materials tested were found to be noninflammable.

Table IV. *1000-cycle harmonic distortion data (measurements are decibel range down from 1000-cycle component).*

Material Group	Lateral Recording Reproducer *A*, Steel Needle		Vertical Recording Reproducer *C*, Diamond Stylus	
	2000 Cycles	3000 Cycles	2000 Cycles	3000 Cycles
1	6–27	14–36	21–36	28–39
2	12–25	20–30		
3	22–24	27–32	17–23	Noisy
4	17–24	27–35	14–22	27–34
5	17–26	23–26	15–21	27–40
Lateral Cut Phonograph Record	20–30	22–34		
Vertical Cut Electrical Transcription			28–34	35–41

Humidity Tests

All of the materials tested were subjected to a humidity test at 95°F and 90 percent relative humidity for 24 hours. The materials of group 1 showed a surface noise increase of from 0 to 5 db in this test. The coated metal disks showed inappreciable warping but the thin disks of recording material alone showed considerable warping.

The materials of groups 2, 3 and 4 showed little evidence of change in the humidity test except for warping of some materials of group 4, where thin disks of resin were used.

The gelatine materials of group 5 were the most seriously affected in this test. The warping was generally very severe and noise measurements were impossible after the test, even for records where the material had been attached to metal disks. For the latter, severe wrinkling of the recording material occurred which caused regional separation from the metal base.

Time did not permit an aging study of these materials which would correspond to storing them as recorded disks for long periods of time. For some uses very little change in the record is tolerable even over a period of years.

Conclusion

The measurements taken show that considerable progress has been made in the field of direct recording materials in the last few years and that many of these disks are capable of reasonably satisfactory recording if used with care. They have been shown to have a life which should be adequate for many uses. The newer disks do not, in general, have the life of good molded records so that their uses may often be considered as complementary to molded records. The new records are thin and light in weight and are therefore easily shipped and stored. Many of them are not easily damaged by ordinary handling, storing or shipping. These newer direct recording materials will therefore probably greatly add to the rapidly growing use of recording.

In order to give an aural demonstration of these recording materials, test records of speech and music have been prepared using both lateral and vertical recording. The comparisons on these test records have been entirely made by re-recording from other records to vertical cut waxes which in turn have been processed to give the molded records which were played. It was possible to do this without appreciable change in the test reproduction because measurements showed the vertical cut molded records to have a wider frequency and volume range and less distortion than any of the materials tested. A 7000-cycle low pass filter was used in the reproducing circuit and corresponds to the upper frequency cut-off used in recording. This was done to reduce surface noise and because practically all of the materials showed very little response above 7000 cycles. Low pass filters having cut-off frequencies of 3200 cycles and 5500 cycles were also available to be connected into the circuit if some in the audience wished to hear a more limited frequency range with, of course, correspondingly less surface noise.

The Principles Underlying the Tuning of Keyboard Instruments to Equal Temperament

G. F. Herrenden Harker
Department of Physics, University College, Cardiff, Wales
(Received June 29, 1936)

KEYBOARD instruments are virtually restricted to the production of a fixed series of notes corresponding to the sequence of digitals on the clavier. It is a simple matter of calculation to allot a definite frequency to each term in this series on theoretical grounds in accordance with some prescribed scheme. The practical problem of adjusting the frequency of each primary sound generator, whether string, reed or pipe, so that, when actuated by the digital mechanism, it shall produce a note of precisely the pitch assigned is a much more intricate undertaking.

Executants upon the earlier stringed keyboard instruments usually undertook their own tuning, and, as it was almost impossible to present a program of any length without allowing intervals for retuning, they had considerable scope for acquiring proficiency. Advances in design, coupled with the introduction of improved materials, have stabilized tuning to such an extent that it has become customary for their successors to delegate these adjustments of their pianofortes to professional tuners who attend to them periodically. The fact that a performer today has been relieved of the necessity of acquiring the technical skill needed to tune his instrument is no valid reason why he should remain in ignorance of the essential problems involved in the tuning process.

The tuning of well-defined intervals, such as octaves or natural fifths, is a relatively simple matter to a musical ear, while the random mistuning of any interval is even easier, and can be perpetrated without any "ear" at all. Paradoxical though it appears at first sight keyboard tuning would be much more readily accomplished if the process really involved tuning. What is actually demanded, and what is quite difficult to achieve, is to realize with exactitude a definite minor degree of mistuning in the sense of a slight, but accurately controlled, departure from the natural interval relationships. This calls for an ear which is scientifically trained rather than musically developed. Apart from the mixture ranks on an organ the only intervals which require tuning in the strict sense are unisons and octaves.

It might seem that the simplest, and most rational, method would be to employ a chromatic set of twelve forks or pitch pipes, one adjusted to reproduce each frequency between tuning A and the G sharp above it, to tune the respective notes to unison with the corresponding standard sources and then, having laid the bearings in this way, to finish the tuning through in octaves. Such a scheme does not appear to be viewed with favor, probably on account of a suspicion that the standard pitches so obtained might not prove sufficiently dependable under ordinary working conditions. Presumably, however, this set of twelve forks would not be subject to any circumstances more detrimental to their stability than is the single standard fork which forms part of every tuner's equipment, though it may be remarked in this connection that the number inscribed on the latter does not invariably represent its actual frequency. Even a good fork, which has been carefully adjusted by a reputable maker at the outset of its career, needs to have its frequency checked from time to time. At all events, although it may be difficult to concede that any great accuracy in tuning is likely to be secured by some of the methods actually in vogue, the principles upon which they rest are relatively simple and scientifically impeccable. Unfortunately, few acoustical texts devote any consideration to this strictly relevant subject.[1] The technical handbooks too often give no hint as to how their practical prescriptions have been arrived at, and not many tuners possess the necessary acoustical equipment to appreciate, or

[1] A notable exception in this, as in so many other acoustical matters, is H. Bouasse (*Acoustique Générale*, p. 216 *et seq.*). The subject is also clearly treated from a practical standpoint in William Braid White's, *Modern Piano Tuning and Allied Arts.*

impart, what is behind the routine methods they employ.[1a]

In view of this it is proposed to give a brief account of the theoretical basis underlying the tuning methods in current use. Though special reference is made to the pianoforte for illustrative material the methods developed are in essence equally applicable to any keyboard instrument, provided it is capable of generating sounds of moderate duration, which are reasonably rich in harmonic content.

Before it is possible to specify the frequencies which should characterize the succession of notes obtainable from a keyboard instrument, a definite frequency must be allotted to some particular note within its compass. That usually selected is the A above middle C, "tenor" or "tuning" A as it is often called, and denoted hereafter as A_3. In scientific circles the frequencies of the various C's are invariably represented by integral powers of 2 and A_3 has the frequency 430.539_0 sec.$^{-1}$, but in the musical world no international agreement upon this important point has ever been secured, and at least three standards of pitch are encountered. In France the "diapason normal," fixed in 1859 and subsequently adopted in 1885 by the Congress of Vienna, assigned to tuning A the frequency 435 sec.$^{-1}$. In the New Philharmonic Pitch (1896), current in England, the frequency of tuning A is 439 sec.$^{-1}$. The Stuttgart Congress of Physicists in 1834 selected for tuning A the frequency 440 sec.$^{-1}$. and this same value was adopted in 1925 by the American Federation of Musicians and Music Industries. All these values, of course, carry with them an appropriate temperature specification, and instruments designed to realize these frequencies must be maintained at the temperature indicated when generating them.

As the interval between the first and third of these standard musical pitches only amounts to a fifth of a semitone, while that between the second and third is about a twenty-fifth of a semitone it seems a needless complication to retain all three. Moreover, as will appear later, the numerical differences involved in the respective tuning schemes rarely, if ever, exceed the limits of experimental error. In any case, however, it is only after the adoption of some such standard of frequency that it becomes possible to evaluate the frequencies of the remaining notes within the compass of the keyboard.

As is well known the distinctive musical character associated with any particular interval is governed by the ratio of the frequencies of the two notes which together constitute that interval.[2] Hence when the frequency of one particular note, e.g., A_3, is given, and the frequency ratio defining the interval between it and any other note is known, the frequency of the latter can be immediately ascertained. In the simplest musical interval apart from unison, the octave, the frequency of the upper note is precisely double that of the lower note. Thus, in terms of A_3 as standard, the frequencies of all the A's within the compass of the keyboard can be at once determined by multiplying, or dividing, this frequency by integral powers of two. The special importance of the octave in the present connection results from the fact that it serves as the foundation interval in every temperament or system of tuning.

The next step is to subdivide these octaves into the smaller intervals demanded by musical usage by the interpolation of a limited number of notes of intermediate pitch. Now there are two obvious methods of partitioning an octave:

(1) to include the maximum possible number of natural intervals,

(2) to contain the minimum requisite number of equal intervals.

In the former, corresponding to just intonation, the intervals between consecutive notes are of unequal magnitude. Freedom of transposition is not possible without the provision of many more notes than the minimum needed to satisfy the normal requirements of classical music. In fact, unless such supernumerary notes are available, remote keys are rendered so offensive to the ear as to be unusable. In the latter, which is that of equal temperament, the maximum sacrifice of

[1a] It is therefore a pleasure to acknowledge my indebtedness to discussions at the pianoforte keyboard with an expert tuner, Mr. W. G. C. Bellingham.

[2] Intervals in tuning are checked objectively by sounding the notes concerned simultaneously, and making appropriate adjustments to secure beating of specified (including zero) frequency. Hence the necessity of envisaging subjective pitch changes due to causes recently explored by Harvey Fletcher (J. Acous. Soc. Am. **6**, 59–69 (1935)) does not arise.

harmony is made in order to secure unrestricted transposition with the minimum permissible number of notes to the octave. Though it violates the basic scientific criterion of harmony—the coincidence of harmonics in notes having different fundamentals—it does enable modulation to any key to be accomplished with the provision of no more than twelve notes to the octave. On this system the unit interval is the equitonic semitone, twelve of which make up the octave. In it none of the intervals within the octave are just, some, as for instance the thirds and sixths, depart so markedly from the corresponding natural intervals that they are hideously rough. Since, however, we get practically no opportunity of listening to intervals which have not been thus tempered, or tampered with as the purist quite legitimately contends, our ears have gradually become immune to their defects. Equal temperament, or what currently passes for it, is adopted on all keyboard instruments and is imposed upon all other instruments which play in their company. Though admittedly a compromise it is a workable one. It operates at the cost of a certain degree of dissonance, which is perceptible, but which has proved, in practice, to be tolerable. In spite of its shortcomings it was sponsored by the great Bach, against whom a charge of tonal insensitiveness could hardly be leveled. And finally no system which makes partial concessions to either basic requirement has proved itself more generally acceptable. The rough approximations to equal temperament which result from inefficient tuning are certainly undesirable.

All ordinary keyboard instruments are, then, actually, or ostensibly, tuned in conformity with the equitonic system in which the unit interval is the equitonic semitone. As twelve of the latter are included in each octave, the frequency ratio characterizing the equitonic semitone will be the twelfth root of two. Hence the successive multipliers which, operating in turn upon any number n, will give the frequencies of the notes included in the octave range between the frequency limits n and $2n$ in the chromatic scale of equal temperament, are, respectively, $2^{0/12}$, $2^{1/12}$, $2^{2/12}$, $\cdots$, $2^{12/12}$. It was deemed advisable to calculate these frequency multipliers with an accuracy in excess of that usually given, or obtainable with the aid of seven-figure logarithms. Since tables of logarithms carried to a greater number of figures are not generally available, the simplest method of procedure is to start with the values of $2^{\frac{1}{2}}$ and $2^{\frac{1}{4}}$, as tabulated, for example, on page 207 of the new edition of *Barlow* to 16 places, to extract the square root of $2^{\frac{1}{2}}$ and then to divide $2^{\frac{1}{2}}$ by the value of $2^{\frac{1}{4}}$ so obtained. The resulting value for $2^{1/12}$ is 1.05946309, and from it values for the remaining multipliers are readily derived. They are contained in Table I.[3]

Eight figures have been retained because in multiplying by numbers of this magnitude it is indispensable to employ adequate assistance, and as Peters' *New Calculating Tables* (Walter de Gruyter and Co., 1924) are extremely convenient they have been used throughout. With them it is just as easy to multiply 87, or 44, by eight figures in two groups of four figures each, or 439 by four groups of two figures each as to rest content with one less figure. And, obviously, the retention of the eight figure increases the reliability of the last figure included in the numerical values of the frequencies of the notes required in connection with the tuning schemes which follow. These frequencies are set out in Table II, and have been derived in the manner indicated. From such a table the frequency of any note upon the keyboard can be deduced immediately since the note itself must be one or more octaves removed from some tabular entry.

We now know what the actual frequency of any, and every, note on a keyboard tuned to any one of the three standard musical pitches should be. But it is another, and a much more difficult, matter to tune the corresponding instrument so

[3] It may not be inappropriate to direct attention to two errors in sources likely to be widely consulted: (1) the report of the Standardization Committee (J. Acous. Soc. Am. **2**, 319 (1930)) gives $2^{11/12}=1.883$ instead of 1.888; (2) in the *Smithsonian Physical Tables* (8th Edition, p. 192) 1.05926 appears in place of 1.05946. It may also be noted that the nth order differences of the tabular entries are given by $2^{0/12}(2^{1/12}-2^{0/12})^n$, $2^{1/12}(2^{1/12}-2^{0/12})^n$, etc., so that, even using approximate values for $2^{1/12}$, $2^{2/12}$, etc., it is possible to check the higher order differences. Thus, for example, the fifth-order differences are

$$1.000(0.05946)^5 = 7.4\times10^{-7}$$
$$1.060(0.05946)^5 = 7.9\times10^{-7}$$
$$1.122(0.05946)^5 = 8.3\times10^{-7}$$
$$1.189(0.05946)^5 = 8.8\times10^{-7}$$
$$1.260(0.05946)^5 = 9.4\times10^{-7}$$
$$1.335(0.05946)^5 = 9.9\times10^{-7}$$
$$1.414(0.05946)^5 = 10.5\times10^{-7}$$
$$1.498(0.05946)^5 = 11.1\times10^{-7}.$$

TABLE I.

1.0	00	00	00
1.0	59	46	31
1.1	22	46	20_5
1.1	89	20	71
1.2	59	92	10_5
1.3	34	83	98_5
1.4	14	21	36
1.4	98	30	71
1.5	87	40	10_5
1.6	81	79	28
1.7	81	79	74
1.8	87	74	86
2.0	00	00	00

that depression of any particular digital shall elicit a note of precisely the calculated frequency. It is customary to work to a single pitch standard —that required to check and set the A_3—and this is usually provided by a tuning fork, though occasionally a pitch-pipe is employed. The technique of tuning in equal temperament relative to this is based upon an aural estimation of beat rates, so that the phenomenon of beating is the indispensable auxiliary to the whole tuning process.

As is well known whenever two tones, which differ but slightly in pitch, are sounded simultaneously beats arise having a frequency which is numerically equal to the difference between the frequencies of the two contributory tones. This beat frequency, when not too great, can be determined by direct count over a timed interval. For this purpose three methods have been advocated.

In the first, the number of beats occurring in a fixed time interval is noted. Since, however, it is exceedingly difficult to estimate fractions of a beat at all precisely, this method, though often recommended, is unscientific and unsatisfactory. In the second, the time corresponding to the completion of a definite number of beats is taken by means of a properly regulated stopwatch on which seconds are subdivided into fifths, or, preferably, tenths. In the third a metronome is adjusted to tick in unison with the beats. When the upper edge of the cursor is aligned on a particular scale reading the latter nominally records the number of ticks given by the metronome per minute. However, since the graduation is seldom, if ever, correct, it is essential to calibrate the instrument against a standard timekeeper. Moreover it is by no means easy to synchronize the metronome ticks with the beat maxima. The ticking of the Maëlzel type is apt to mask the beating, so that it may be preferable to employ the simpler silent metronome which is gravity controlled instead of clockwork actuated. Failing this a home-made, simple pendulum has been proposed. Synchronization is then secured by the use of the eye in conjunction with the ear.

The second of these three alternatives would appear to be the most rational, and offer the best prospect of accuracy in checking a beat frequency. Since, however, the central problem in tuning is to realize a definite prescribed beat frequency rather than to determine a preexisting unalterable one, an instrument like the metronome, which can be set to reproduce the actual frequency to which the beats have to be adjusted, has obvious advantages. Apart from the drawbacks already indicated the graduations on existing types of metronome are so widely spaced that the cursor has often to be set intermediarily by eye. There is, of course, no reason why a more suitable metronome should not be specially designed for this purpose.

The accuracy obtainable in realizing a prescribed beat frequency will depend not only upon which of the three methods outlined above is adopted for timing but also upon the actual beat rate to be timed, and the interval during which the beating remains audible. If the beat rate is too slow difficulty is experienced in locating the sound maxima and minima with precision,

TABLE II. *Frequencies of notes employed in calculating beat frequencies used in tuning to equal temperament with $A_3 = 435$ sec.$^{-1}$, $A_3 = 439$ sec.$^{-1}$ and $A_3 = 440$ sec.$^{-1}$, respectively.*

$G_2\sharp$ $A_2\flat$	205.292_7	207.180_4	207.652_4
A_2	217.500	219.500	220.000
$A_2\sharp$ $B_2\flat$	230.433_2	232.552_2	233.081_9
B_2	244.135_5	246.380_4	246.941_6
C_3	258.652_6	261.031_0	261.625_6
$C_3\sharp$ $D_3\flat$	274.032_8	276.552_7	277.182_6
D_3	290.327_7	292.997_4	293.664_8
$D_3\sharp$ $E_3\flat$	307.591_4	310.419_9	311.127_0
E_3	325.881_8	328.878_4	329.627_6
F_3	345.259_7	348.434_5	349.228_2
$F_3\sharp$ $G_3\flat$	365.789_9	369.153_5	369.994_4
G_3	387.540_9	391.104_5	391.995_4
$G_3\sharp$ $A_3\flat$	410.585_3	414.360_8	415.304_7
A_3	**435.000**	**439.000**	**440.000**
$A_3\sharp$ $B_3\flat$	460.866_4	465.104_3	466.163_8
B_3	488.271_0	492.760_8	493.883_3
C_4	517.305_1	522.061_9	523.251_1
$C_4\sharp$ $D_4\flat$	548.065_7	553.105_3	554.365_3
D_4	580.655_3	585.994_7	587.329_5

though the likelihood of a miscount is remote. If the beat rate is too rapid the maxima and minima are sharply defined, but the reliability of the count is reduced. Optimum conditions correspond to between one and two beats per second and a persistence of the beating pair of at least fifteen seconds to enable a reasonably prolonged, and sufficiently definitive, count to be undertaken. The question of the time during which beating remains audible does not arise with the organ or harmonium since notes on them can be sustained indefinitely. Provided the requisite trouble is taken these instruments really can be tuned to equal temperament, and the harmonium, whose reeds are comparatively little affected by ordinary temperature excursions, is capable of providing a more satisfactory approximation to equal temperament than any other keyboard instrument. Here there is some justification for giving, as is sometimes done, the number of beats that should be heard per minute when laying the bearings in equal temperament. Any idea of applying the same procedure to the pianoforte is fantastic. Even in the extreme bass a pianoforte note will barely persist for a minute; in the central region of the keyboard the interval during which beat counting can be undertaken varies between ten and twenty seconds, and a note in the extreme treble becomes inaudible in less than half this time. When, further, it is remembered that the harmonics, between which the beats actually occur, decay even more rapidly than the corresponding fundamentals the very real difficulty involved in the reliable estimation of even favorable beat rates on the pianoforte becomes patent.

If two notes, as opposed to two tones, are sounded simultaneously, beats may arise even though their fundamentals lie outside beating range of one another. Such beats occur between individual harmonics, which figure as constituent tones in the structure of the respective notes. Since a musical interval is specified by a frequency ratio, and a beat frequency by a frequency difference, it follows that two notes separated by the same interval will give different beat rates, and so produce different aural impressions, in different regions of the keyboard. Thus, for a given interval, the beat frequency will be less in the bass than in the treble. The beat effect is particularly striking when the components of the beating pair are of approximately equal intensity since, then, the sound minima reduce practically to silences. An attempt should be made to obtain equality of intensity between the members of the beating pair whenever accuracy in beat counting is aimed at. It is not always possible to realize this optimum condition since the harmonics of different orders enter into the constitution of notes from different sources in different relative proportions. A certain amount of adjustment is feasible on the pianoforte since the notes concerned can be struck with unequal vigor. No measure of compensation by graduation of the blow is effective on the organ or harmonium. Furthermore the presence of the fundamentals, and of harmonics other than those between which beating occurs, masks what beating there is to some extent and renders its precise evaluation more difficult.

Suppose, for example, that the A_3 on a keyboard has been brought into unison with some standard pitch given, for instance, by a 439 fork, then the note D_3, an equitonic fifth below it in pitch, must be so tuned that its frequency is 292.997_4. Now these two notes will include as harmonic ingredients some, or all, of the following frequencies:

A_3	439	878		1317	1756
D_3	292.997_4	585.994_8	878.992_2	1171.989_6	1464.987_0	1757.984_4.

Hence the second harmonic of A_3 will beat with the third harmonic of D_3, and the corresponding beat frequency will be 0.992 sec.$^{-1}$. Thus, when D_3 has been correctly tuned in equal temperament with respect to this particular A_3, it will give 10 beats in 10.1 sec., when sounded in company with it. In addition the fourth harmonic of A_3 will beat with the sixth harmonic of D_3 at a rate double the above (10 beats occupying 5 sec.), but, in view of the superior rank and proportionately diminished prominence of this pair, such beats are unlikely to prove obtrusive. Higher harmonics than those listed will almost certainly be insignificant in comparison.

Suppose, however, that the frequency of the D_3 were adjusted to be exactly two-thirds that of this same A_3, i.e., equal to $292\frac{2}{3}$ sec.$^{-1}$. Then D_3 would be a natural fifth below A_3 in pitch and the harmonic scheme would become:

A_3		439	878	1317		1756
D_3	$292\frac{2}{3}$	$585\frac{1}{3}$	878	$1170\frac{2}{3}$	$1463\frac{1}{3}$	1756.

The second and fourth harmonics of A_3 are now coincident in pitch with the third and sixth harmonics of D_3, respectively. As no beats can arise between these unisons the two notes will, on this score, present an unblemished concord when sounded together. A comparison of the fundamental frequency of the A_3 with those of the tempered and natural D_3's indicates that the equitonic interval of the fifth is smaller than the corresponding natural interval, though the extent of the departure is so slight that it is impossible to distinguish between them when used melodically, i.e., in circumstances in which beats as an accessory guide are excluded. Employed harmonically, beats at once reveal the discrepancy, small though it is.

Accordingly, a rule for tuning D_3 an equitonic fifth below A_3 can be formulated as follows: (1) tune D_3 a natural fifth below A_3 so that no beats are perceptible, and the effect of sounding them together is smooth; (2) progressively sharpen the D_3 (thus contracting the interval between it and A_3) until, when sounded in company with the original A_3, 10 beats can be counted in 10.1 seconds.

It is invariably safer to obtain the natural interval first, and then mistune it to secure the corresponding equitonic interval as this obviates any uncertainty as to the direction in which its alteration should proceed. Since the differences between the natural and equitonic intervals are usually quite small it is difficult, otherwise, to be sure whether a particular beat rate denotes that the interval obtained exceeds, or falls short of, the corresponding natural interval. Moreover appreciation of the natural intervals, and familiarity with the tonal background in the absence of beats, materially assists in checking the frequency of the beats when these are superimposed upon it.

It will be clear from the above example that, when the interval between two notes having frequencies n and n' is an equitonic fifth, the frequency of the beats that arise will be given by $3n-2n'$. Its value will accordingly depend upon the numerical values of n and n', and no two intervals of the same degree will beat at exactly the same rate. Since $3n-2n'>0$, and $=2n(3/2-n'/n)$, it follows that $3/2>n'/n$, in other words that the natural fifth is larger than its equitonic counterpart as has already been pointed out.

Since any scheme for tuning must embrace individually each of the thirteen notes comprised within an octave at least twelve intervals will require to be beat tuned. The two intervals almost exclusively used for this purpose are the fifth and fourth. They contain, respectively, 7 and 5 semitones, and, as 7 and 5 are the only numbers under 12 (apart from 1 and 11 irrelevant in this connection) which are prime with respect to 12, it is clearly possible to include each note within the octave by proceeding in fifths and fourths. Some schemes necessitate octave tunings in addition, but these may be assumed to involve a negligible error since octaves are tuned dead. In the same way the starting note may be assumed correct since it is unison tuned to the pitch standard. The test octave, within which the bearings are laid, extends from A_2 to A_3 and occupies the central section of the keyboard compass. Over this region the sound persistence on the pianoforte is reasonable. With a persistence of fifteen seconds, and assuming the possibility of detecting the presence of a single beat in this interval, unison tuning in the vicinity of A_2 is possible to within some five-thousandth of a semitone and, in the vicinity of A_3, to approximately half of this. Moreover the corresponding beat rates of the fifths and fourths between these limits are suitable for control by direct count.

If an attempt is made to lay the bearings an octave higher the respective beat frequencies are doubled, which enhances the difficulty of counting, while the more rapid extinction of the sounds lessens the interval over which counting is practicable. Working an octave lower the beat frequencies would be halved, and the beats rather too slow to be fixed with precision, though the sound persistence would be increased. Some compromise is accordingly inevitable, and that which experience has shown to be most advantageous is the one adopted.

Over the range in question it is possible to time at least 10 beats on the pianoforte and, if t_{10} is the corresponding time in seconds, $t_{10}=10/\eta=10/(3n-2n')$, where η denotes the

beat frequency and n and n' are the frequencies of two notes an equitonic fifth apart in pitch. If m represents the metronome setting which synchronizes with this beat frequency η, then $m=60\eta=60(3n-2n')$. In Table III the values of η have been evaluated for the series of fifths employed in the process of laying the bearings on a keyboard in equal temperament for each of the three standard musical pitches $A_3=435$, 439 and 440 sec.$^{-1}$.

From the tabulated values of η it will be evident that it is amply sufficient, in practice, to adopt as applicable to any pitch between 435 and 440 the corresponding values of t_{10} and m set out in the last two columns. The discrepancy involved in t_{10} does not anywhere exceed 0.1 of a second, which is safely within the limits of experimental error inevitable in beat timing even with a well regulated stopwatch reading to tenths of a second. Moreover beats afford such an extremely sensitive means of pitch discrimination that an error of this order is insignificant in terms of frequency and interval. Consider, for example, the fifth D_3-A_3, the frequencies of the respective notes being n and n' of which the latter may be assumed correct since it has been unison tuned to a fork. Then, since $t_{10}=10/(3n-2n')$, the error on n corresponding to an error δt_{10} on t_{10} will be given by $\delta n=-(10/(3t_{10}^2))\delta t_{10}$, so that, if $t_{10}=10.1$ sec. and $\delta t_{10}=\pm 0.1$ sec., $|\delta n|=0.0033$. But, as the interval between two neighboring notes of frequencies n and $n\pm\delta n$ is equivalent to $17.3|\delta n|/n$ of a semitone, the error of 0.0033 on a frequency in the vicinity of $n=293$ sec.$^{-1}$ will only amount to some two ten-thousandths of a semitone.

TABLE III.

Fifths n	Fifths n'	$A_3=435$ sec.$^{-1}$ η	$A_3=439$ sec.$^{-1}$ η	$A_3=440$ sec.$^{-1}$ η	In Practice for Any Pitch Between 435 and 440 t_{10}	m
$A_2\flat(G_2\sharp)$	$-E_3\flat(D_3\sharp)$	0.695	0.701	0.703	14.3	42
$A_2(B_2\flat\flat)$	$-E_3(F_3\flat)$	0.736	0.743	0.745	13.5	45
$B_2\flat(A_2\sharp)$	$-F_3$	0.780	0.788	0.789	12.7	47
$B_2(C_3\flat)$	$-F_3\sharp(G_3\flat)$	0.827	0.834	0.836	12.0	50
C_3	$-G_3$	0.876	0.884	0.886	11.3	53
$C_3\sharp(D_3\flat)$	$-G_3\sharp(A_3\flat)$	0.928	0.936	0.938	10.7	56
D_3	$-A_3$	0.983	0.992	0.994	10.1	60
$E_3\flat(D_3\sharp)$	$-B_3\flat(A_3\sharp)$	1.041	1.051	1.053	9.5	63
E_3	$-B_3$	1.103	1.114	1.116	9.0	67
F_3	$-C_4$	1.169	1.180	1.182	8.5	71
$F_3\sharp$	$-C_4\sharp$	1.238	1.250	1.253	8.0	75
G_3	$-D_4$	1.312	1.324	1.327	7.6	79

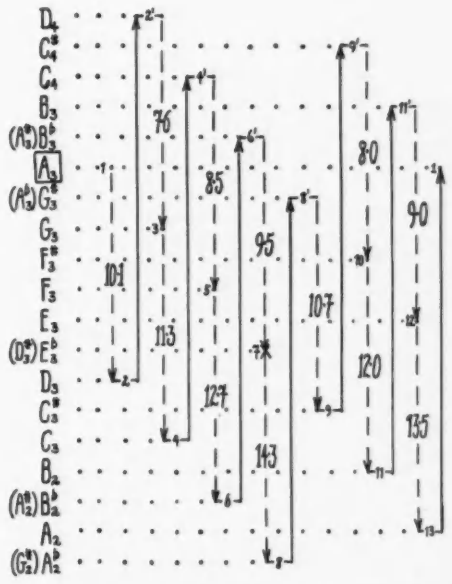


FIG. 1. Scheme employing fifths and octaves and using beats to lay the bearings in equal temperament. Figures give time in seconds of 10 beats for any pitch between $A_3=435$ sec.$^{-1}$ and $A_3=440$ sec.$^{-1}$. All fifths to be contracted. All octaves tuned dead.

Values of m are, for the same reason, given to the nearest integer since to attempt fractional settings on the scale of the metronome is unlikely to prove profitable, and as an error of a unit in m likewise only corresponds to a few ten-thousandths of a semitone over the test octave.

On the basis of this table it becomes possible to evolve a scheme for laying the bearings in equal temperament. Starting from A_3, tuned to the appropriate standard fork, the tuner proceeds by descending fifths and ascending octaves contracting all fifths relative to the corresponding natural fifths to give 10 beats in t_{10} seconds, as specified in Table III, and tuning all octaves dead. If these operations are conducted in the order exhibited diagrammatically in Fig. 1, they will, if carried out successfully, accomplish the tuning in equal temperament of the thirteen notes comprised in the octave interval A_2 to A_3. The tuner, having thus partitioned the octave into twelve equitonic semitones, should, at the end, arrive back at his starting point. The proof of the procedure is, in point of fact, that the final ascent of an octave from A_2 gives a note A_3 identical in pitch with the initial A_3 as set by the fork.

To avoid accumulation of errors, which,

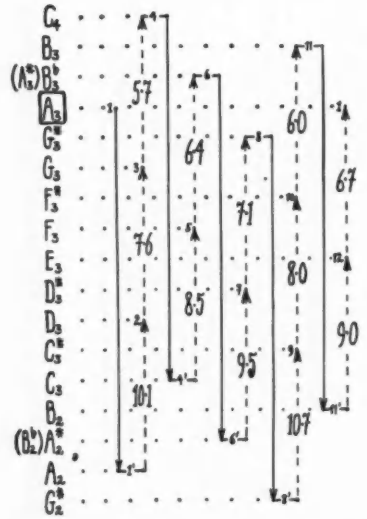


FIG. 2. Scheme employing fourths and octaves and using beats to lay the bearings in equal temperament. Figures give time in seconds of 10 beats for any pitch between $A_3=435$ sec.$^{-1}$ and $A_3=440$ sec.$^{-1}$. All fourths to be expanded. All octaves tuned dead.

though individually imperceptible, may, at the conclusion of such a series of operations, have attained unacceptable proportions, a more prudent procedure would be to start from A_3 and tune in descending fifths and ascending octaves as far as the half-way point (the note $D_3\sharp$ marked with a cross), and then to work backwards from the other end, tuning from A_3 by descending octaves and ascending fifths until a half-way point is again reached, which must be the same note $D_3\sharp$. It must be remembered that, to secure the requisite contraction of the respective natural fifths, the final note in all intervals except octaves will require sharpening in the first half of such a procedure, while, in the second half, the final note in all intervals other than octaves will need flattening.

The method just outlined thus employs twelve descending, or ascending, equitonic fifths and seven ascending, or descending, perfect octaves in laying the bearings over the nineteen note range between $G_2\sharp$ and D_4, inclusive. It is, however, by no means the only one that can be devised. It is, for example, possible to cover the range including the seventeen notes between $G_2\sharp$ and C_4 by means of twelve ascending, or descending, equitonic fourths and five descending, or ascending, perfect octaves in the manner set out diagrammatically in Fig. 2.

The beat frequency corresponding to the interval separating two notes, of frequencies n and n', an equitonic fourth apart in pitch is readily seen to be given by $3n'-4n$. Since this expression, which may be written in the form $3n(n'/n-4/3)$, is positive it follows that $n'/n>4/3$, i.e., that the equitonic interval of the fourth exceeds the corresponding natural interval. Hence, in this case, the natural interval must be expanded to an extent indicated by the calculated beat frequency in order to obtain the equitonic interval. The values of η, t_{10} and m for the sequence of fourths involved in such a scheme are set out in Table IV.

It is clear that the beat frequencies given by equitonic fourths do not exceed those given by equitonic fifths over the central region of the keyboard sufficiently to render their timing notably less reliable. The fact that it is easier for a musically trained ear to tune a natural fifth than a natural fourth is entirely irrelevant in this connection since the ear is simply employed as a counting mechanism to check a beat rate. It is not appealed to in its musical capacity to pass judgment on the exactness, or even the degree of inexactness, of any particular natural interval relationship. The only region where a sense of pitch is likely to prove an asset is in the extreme treble, where the sounds are of very short duration, the harmonics relatively inconspicuous, and where, actually, it is customary to tune slightly sharp as otherwise arpeggios convey an impression of being out of tune.

In practice it is more usual to employ both fifths and fourths in laying the bearings, i.e.,

TABLE IV.

Fourths n	n'	$A_3=435$ sec.$^{-1}$ η	$A_3=439$ sec.$^{-1}$ η	$A_3=440$ sec.$^{-1}$ η	In Practice for Any Pitch Between 435 and 440 t_{10}	m
$G_2\sharp$	$-C_3\sharp$	0.928	0.936	0.938	10.7	56
A_2	$-D_3$	0.983	0.992	0.994	10.1	60
$A_2\sharp(B_2\flat)$	$-D_3\sharp(E_3\flat)$	1.041	1.051	1.053	9.5	63
$B_2(C_3\flat)$	$-E_3(F_3\flat)$	1.103	1.114	1.116	9.0	67
C_3	$-F_3$	1.169	1.180	1.182	8.5	71
$C_3\sharp(D_3\flat)$	$-F_3\sharp(G_3\flat)$	1.238	1.250	1.253	8.0	75
D_3	$-G_3$	1.312	1.324	1.327	7.6	79
$D_3\sharp(E_3\flat)$	$-G_3\sharp(A_3\flat)$	1.390	1.403	1.406	7.1	84
E_3	$-A_3$	1.473	1.486	1.490	6.7	89
F_3	$-B_3\flat(A_3\sharp)$	1.560	1.575	1.579	6.4	95
$F_3\sharp$	$-B_3$	1.653	1.668	1.672	6.0	100
G_3	$-C_4$	1.752	1.768	1.772	5.7	106

to combine to some extent the two schemes already outlined, and which rely exclusively upon one or the other. One such scheme is exhibited diagrammatically in Fig. 3. It possesses an advantage over the two previous schemes in involving somewhat fewer operations since the bearings are laid by means of six descending, or ascending, fifths and six ascending, or descending, fourths without going outside the octave A_2 to A_3. The relevant numerical data are included in Tables III and IV, from which they may be extracted.

Whatever the scheme actually adopted it is advisable to institute some system of cross checking to verify the various stages of the work as it progresses. One useful test depends upon the fact that, starting from any note, the beat frequency corresponding to the interval of an equitonic fifth above it must be equal to the beat frequency corresponding to the interval of an equitonic fourth below it, while the interval separating the extreme notes must necessarily be a perfect octave.

Let n be the frequency of the initial note, n' the frequency of the note an equitonic fifth above it in pitch, and n_1 the frequency of the note an equitonic fourth below it in pitch. Then, since the combined interval corresponds to a perfect octave, $n'=2n_1$. But, since the beat frequency corresponding to the fifth is given by $3n-2n'$ and that corresponding to the fourth by $3n-4n_1$, these must be equal since $n'=2n_1$.

This rule is of service for checking octaves in the extreme bass of the pianoforte from which it is often difficult to exclude all trace of beating since the overspun wires are liable to furnish false beats on their own account. Illustrations of its application are afforded in Tables III and IV, from which it is seen that the beat frequencies corresponding to the series of fifths $D_3\sharp-A_3\sharp$, E_3-B_3, F_3-C_4, $F_3\sharp-C_4\sharp$, G_3-D_4 are, respectively, identical with those corresponding to the series of fourths $A_2\sharp-D_3\sharp$, B_2-E_3, C_3-F_3, $C_3\sharp-F_3\sharp$, D_3-G_3. The converse of this rule does not, however, hold. If the intermediate note has a frequency $n\pm\delta n$, instead of the frequency n required in equal temperament, the respective beat frequencies will be $3(n\pm\delta n)-2n'$ and $3(n\pm\delta n)-4n_1$ and will still be equal so long as

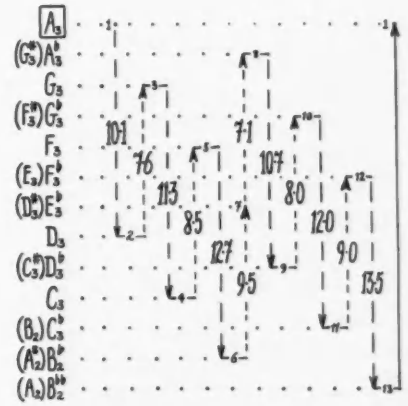


FIG. 3. Scheme employing fifths and fourths and using beats to lay the bearings in equal temperament. Figures give time in seconds of 10 beats for any pitch between $A_3=435$ sec.$^{-1}$ and $A_3=440$ sec.$^{-1}$. All fifths to be contracted. All fourths to be expanded.

$n'=2n_1$, i.e., the two extreme notes stand to one another in true octave relationship.

With the order of the intervals reversed the beat frequency corresponding to an equitonic fourth above a given note will be double the beat frequency corresponding to an equitonic fifth below the same note. The respective beat frequencies being $3n'-4n$ and $3n_1-2n$, it follows, since $n'=2n_1$, that $3n'-4n=2(3n_1-2n)$. Examples are afforded in Tables III and IV, from which it is seen that the beat frequencies corresponding to the series of fourths $D_3\sharp-G_3\sharp$, E_3-A_3, $F_3-B_3\flat$, $F_3\sharp-B_3$, G_3-C_4 are, respectively, double those corresponding to the series of fifths $A_2\flat-E_3\flat$, A_2-E_3, $B_2\flat-F_3$, $B_2-F_3\sharp$, C_3-G_3. Similar rules obviously apply to other intervals and their inversions, e.g., major third and minor sixth, minor third and major sixth, though they are of no particular practical importance in the present connection.

Another test is to verify that the three equitonic major thirds into which the octave is divisible are equal. The beat frequency arising between two notes of frequencies n and n', an equitonic major third apart in pitch, is $4n'-5n$. Since this expression is positive it follows that $n'/n>5/4$, i.e., that the equitonic interval exceeds the corresponding natural interval so that the latter must be expanded to conform with the former. Splitting up the octave A_2 to A_3 into the three major thirds $A_2-C_3\sharp$, $C_3\sharp-E_3\sharp$, F_3-A_3 the respective beat frequencies are given in Table V.

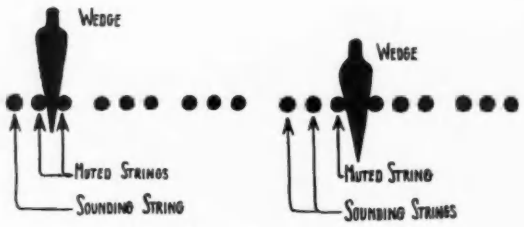
Inspection shows that these beat frequencies are of a different order of magnitude from those encountered in either the fifths or fourths. This, of course, is due to the much greater divergence between the natural and equitonic intervals in the case of the major third. Similar results are found for the minor third and the sixths. Manifestly it is hopeless to expect to be able to count beats at such rates satisfactorily, so that there is no point in giving either the times for ten beats, or metronome indications. All that can reasonably be done is to observe whether the general impression created by the beating increases regularly when the thirds are sounded in the order indicated.

The whole art of tuning to equal temperament along the lines indicated hinges upon a delicate discrimination between, and an accurate determination of, different rates of beating. Not only is the beat method extremely sensitive, it is, at the same time, objective and so independent of any subjective estimate of intervals. In the practical technique of keyboard tuning beats are employed in two ways: (1) the absence of beating between two notes denotes the attainment of unison, or the realization of the octave. If a unison is not true, beating occurs between the fundamentals: if an octave is false, between the second harmonic of the lower note and the fundamental, or first harmonic, of the upper note; (2) the presence of beating at a definite rate indicates that a specified degree of mistuning of the interval between two notes has been secured. Since in equal temperament no interval within the octave agrees exactly with the corresponding natural interval, beating at a definite, calculable, frequency must arise when any such interval is sounded on the keyboard.

Tuning to obviate beating is exemplified in truing up the unisons, an operation which comprises some two-thirds of the whole work on the pianoforte since the majority of its unisons are

TABLE V.

Major Thirds n — n'	$A_3=435$ sec.$^{-1}$ η	$A_3=439$ sec.$^{-1}$ η	$A_3=440$ sec.$^{-1}$ η
$A_2-C_2\sharp$	8.63	8.71	8.73
$C_3\sharp-E_3\sharp$	10.88	10.97	11.00
$F_3(E_3\sharp)-A_3$	13.70	13.83	13.86



(A) (B)

FIG. 4.

double, or triple, strung. Thus in tuning the pianoforte A_3 to the standard fork, a rubber wedge is inserted, as indicated in Fig. 4A, between two of its three strings to put them temporarily out of commission, leaving the remaining member of the trichord free to sound. The tuning hammer is engaged over the square head of the corresponding wrest pin and rotated until, when the fork and string are sounded simultaneously, no beats are heard. The wires are so wound on the tuning pins that clockwise rotation of the hammer increases the tension in them thus raising their pitch, and *vice versa*. "Solid" tuning is impossible without proper "setting" of the pin. If it is wrenched, or "sprung," the string will soon knock out of tune. The wedge can then be withdrawn and reinserted, as indicated in Fig. 4B, between the next pair of strings. This leaves two out of the three strings free to respond to the hammer, and, normally, beats will be heard when the appropriate key is struck. Leaving the original string, already matched to the fork, untouched, the other string is tuned until beating ceases. The wedge is then removed altogether, and the third string brought into unison with the other two.

A similar procedure is followed in octave tuning. First one string is adjusted until no beats are perceptible when it is sounded in company with the previously tuned octave. Then the remaining two strings can be tuned to unison with the string already tuned to the octave.

In laying the bearings the more usual practice is to mute the region concerned by using a long strip of baize inserted "chinese-cracker" fashion, as shown in Fig. 5A, to stop off the outer strings of each trichord. The undamped middle strings are then tuned to equal temperament by ad-

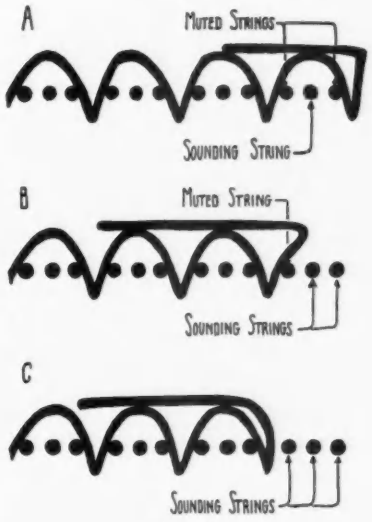


FIG. 5.

justing their tensions until they furnish beats of the requisite frequency when sounded together in pairs in accordance with one of the schemes already prescribed. Subsequently the unisons are established by pulling out the strip one loop at a time, as shown in Fig. 5B and C. Once the test octave has been tuned, the remainder of the tuning proceeds by ascending, or descending, octaves. It is essential that the bearings should be well and truly laid within the test octave since any faults committed there will be propagated, and multiplied up, by transmission throughout the entire compass of the instrument.

The basic principles governing the use of beats in laying the bearings may be summarized as follows:

(1) Ascertain whether the natural interval requires expanding, or contracting, in order to arrive at the corresponding equitonic interval.

(2) Calculate the beat frequency characterizing the particular equitonic interval in question, and verify that its magnitude is such as to permit of accurate determination.

(3) Assuming the note which serves as starting point to be correctly tuned, tune the other note until absence of beating indicates the attainment of the corresponding natural interval.

(4) Make the requisite adjustment upon the final note, sharpening it if it is above the first note in pitch and the interval needs expanding (or if it is below the first note in pitch and the interval needs contracting), and *vice versa*, until the correct beat frequency is obtained.

Only the two intervals of the equitonic fifth and fourth provide beats at a rate suitable for accurate estimation over the test octave, and Tables III and IV show how little their beat frequencies vary with the situation of these intervals within this range. Thus, with $A_3=439$, the beat frequency corresponding to the lowest fifth used in laying the bearings is 0.701, that corresponding to the highest fifth used is 1.324 and, in between them, ten fifths are located having beat frequencies intermediate between these extreme values. If tuning to equal temperament is really to be achieved it is imperative that beats at precisely the frequencies calculated should be secured. It is virtually impossible to control beat frequencies varying within such narrow limits with any approach to precision unless some suitable scientific technique is employed. Tuners practically never attempt to use any timing device to check beat frequencies, and, in fact, the time they are expected to devote to a tuning hardly permits of undue refinement in this respect. In laying the bearings they either aim at contracting the fifths, and expanding the fourths, wherever they occur about a "wave," which, presumably, is the estimated equivalent of a beat a second; or they attempt, even more vaguely, to tune these intervals "slightly," a "trifle" or a "shade" flat or sharp.[4] This implies an effort to rely on the ear as a judge of interval relationships under the worst possible conditions at, or beyond, its limits of discrimination rather than to utilize it simply as a beat counter. When it is not feasible to employ beats as an auxiliary to the tuning process, and regulate their frequency with the requisite accuracy, even a first class tuner is unlikely to realize an equitonic interval on a keyboard instrument with exactitude. As by no means all tuners merit inclusion in this category it is permissible to doubt whether such "trifling" methods can result in anything better than a "shady" substitute for the alleged equal temperament.[4a]

[4] Compare Brinsmead, *History of the Pianoforte*, pp. 147–150.

[4a] Some interesting results of an investigation into the accuracy attained by experienced pianoforte tuners have recently been reported by M. Grützmacher and W. Lottermoser (Physik. Zeits. **36**, 903–12). Irregular deviations up to a fifth of a semitone from the equitonic intervals of the semitone, major third, fifth and octave were found within the range C_3 to C_5. It is somewhat surprising that the discrepancy in the tuning of octaves should turn out to be of the same order as that in tuning other and less well-defined intervals. Closer conformity to equal temperament

That such scepticism is not wholly unfounded is indicated by the fact that different keys have become associated, in the minds of certain musicians, with special peculiarities, variously described as bright, martial, melancholy, etc. Statements such as "just as the key of C major seemed to Mozart the most suitable for the expression of his noblest thoughts, so that of G minor was the vehicle of his most intimate spiritual experience" are commonly encountered. Sharp keys are reputed to be brighter than flat keys, though such a contention is manifestly not of unrestricted validity. Otherwise, a composition in seven sharps should be superlatively bright, and the same composition, played in five flats, distressingly somber in spite of the fact that identically the same notes would actually be played. The fanciful connection of keys with colors in the minds of the hyper-imaginative may be dismissed as irrelevant to the subject of acoustics.

It must not be supposed, however, that all musicians, or even the most eminent among them, subscribe to this view. One composer will maintain in all sincerity that, when he writes a song, he intends it to be performed as written and not transposed at the behest of a vocalist, that, if he had conceived it in another key, he would almost certainly have written quite a different pianoforte part. On the other hand Schubert, one of the most spontaneous composers who ever lived, had so little instinctive conviction concerning the atmosphere created by, and pertaining to, a particular key that he not infrequently transposed his songs, without modifying either the vocal line or the pianoforte accompaniment. He did this, for instance, when assembling some of his finest songs for inclusion in the "Winterreise" cycle.[5] Now, in a song, the prevailing mood is dictated by the words, and some attempt is usually made to mirror this in the music, as is abundantly evident in Schubert's lieder. When, therefore, the original version of "Mut" was written in A minor, and the version finally published in the "Winterreise" set appeared in G minor without further alteration, it can only be concluded that, to Schubert's highly developed musical perception, the keys of A minor and G minor were equally competent to convey an impression of "courage." Apparently he likewise found the keys of D minor and C minor equally "restful," and D minor and B minor equivalent as a means of depicting "loneliness." There appears to be no contemporaneous evidence that Schubert, in making such changes, was considering the human element and arranging these songs to lie within the vocal compass of any individual artist.

Further it is interesting to note that, on the continent during the classical period, the pitch of tuning A rose from about 410 sec.$^{-1}$ in 1700 to about 420 sec.$^{-1}$ in 1810. Taking the mean as 415 sec.$^{-1}$, this is almost exactly a semitone below the present day 440 pitch. Hence, in order to appreciate classical music strictly as its composers intended, it ought to be transposed down a semitone for modern editions. A good deal of vocal music would assuredly prove more grateful both to performers and audiences in this guise, however devastating its impact upon the key-conscious musician.

Assuming a keyboard instrument to be accurately tuned to equal temperament, playing the same composition upon it in two different keys simply means playing it at two different pitches. Though all the intervals involved conserve their magnitudes unchanged, it is, to some extent, true that the same interval does produce a somewhat different aural impression according to its relative situation on such a keyboard. The beat frequency characterizing the interval does, in point of fact, depend upon the absolute pitch of the notes comprising the interval, and the effect of such beating is necessarily more pronounced since the intervals have been tempered. If this represents the whole story it is clear that the change in the characteristic impression produced by a composition should be least marked for a transposition involving the smallest interval shift, since this will cause the least variation in the interharmonic beating. Apparently, however, this is not so. A composition written in A, and transposed to A♭, which simply lowers the absolute pitch of each note through a semitone, the smallest interval on the keyboard, is liable to

was obtained in tuning the 8 ft. Dolce and Principal ranks of an organ. Details are also given of an electrical method of frequency control affording a very close approximation to equal temperament.

[5] I am indebted to my friend Professor G. A. Boutry for drawing my attention to this very illuminating example.

alter in character more radically than when it is transposed to the dominant, which necessitates the raising of each note through the much larger interval of the fifth in pitch, or even when simply played an octave higher or lower. The more intimate relationship that seems to obtain in the latter cases as compared with the former may admit of interpretation on the assumption that keyboard instruments, and the pianoforte in particular, in spite of the most conscientious efforts on the part of the tuner, practically never conform rigorously to equal temperament. In some keys the actual intervals are liable to approach more nearly to natural intervals than to their equitonic counterparts; in others, less closely allied to the tonic, they may depart more than they ought from the natural intervals. Scales which, in the various keys, should reproduce the identical intervals in the same order acquire a certain individual bias, and tend to resemble modes. The procedure for tuning having become more or less standardized, some measure of consistency in these pseudomodal variants of the scale is to be anticipated.

In this connection it is instructive to compare the frequencies of the notes of the major diatonic scale in its equitonic and natural forms taking, for example, $A_3 = 439$ sec.$^{-1}$, as is done below:

	A_2	B_2	$C_3^{\sharp}$	D_3	E_3	$F_3^{\sharp}$	$G_3^{\sharp}$	A_3
Equitonic diatonic major scale.....	219.5	246.38	276.55	293.00	328.88	369.15	414.36	439
Natural diatonic major scale.....	219.5	246.94	274.38	292.67	329.25	365.83	411.56	439
Relative to the natural note the equitonic note is.....	correct	0.04 of a semitone flat	0.14 of a semitone sharp	0.02 of a semitone sharp	0.02 of a semitone flat	0.16 of a semitone sharp	0.12 of a semitone sharp	correct

The greatest divergence between the respective pairs of notes (which occurs on the sixth degree of the scale) is slightly less than one-sixth of a semitone, while the least (which occurs on the fifth degree of the scale) only amounts to just under a fiftieth of a semitone. Minute though this latter interval is, the differentiation between the natural and equitonic interval $A_2 - E_3$ is perfectly possible since, when the former is sounded no beating should be perceptible, while, when the latter is sounded, beats of frequency 0.74 sec.$^{-1}$ should be heard, i.e., 10 beats in 13.5 seconds. If the two E_3's are sounded simultaneously beating at half this frequency, i.e., 0.37 sec.$^{-1}$, would be heard, corresponding to 10 beats in 27 seconds. Such fine limits permit only a slender margin of error and, as errors in tuning tend to be cumulative, the accuracy of the final result will depend, partly upon the sequence adopted in partitioning the test octave, and partly upon the system of cross-checking employed. Usually tuning is carried out relative to an A fork, though a C fork is sometimes used. Assuming consistent adoption of the same tuning scheme to lead to a similar slight deviation from the theoretical equitonic partitioning of the octave, it is not unreasonable to anticipate that some measure of parallelism, as regards general effect, should exist between the key of C on a keyboard tuned to a C fork, and the key of A on a keyboard tuned to an A fork; between the key of G on the former and the key of E on the latter, etc., a possibility which could readily be tested.

If the subjective impression created by a key persists independently of pitch the presumption is that the temperament is unequal. A very pertinent argument in favor of this view, that key color is bound up with the imperfect realization of equal temperament on a pianoforte, is provided by the fact that musicians appear to agree that the modification in musical effect consequent upon transposition is much less marked on the organ or harmonium which, in virtue of their sustaining power, can be made to conform much more closely to equal temperament.

The only alternative interpretation which seems to have been advocated is that playing in certain keys involves more frequent use of the black digitals on the keyboard, that in the pianoforte the hammer blows resulting from a similar attack on a black and a white digital differ perceptibly, and that such action differences, variously distributed throughout the degrees of

the scale, impart a distinctive character to the various keys. Though the raised and shortened levers of the black digitals may favor a somewhat greater facility of attack, the lever arm ratio (2 : 3) is identical for both black and white digitals, so that the back lift corresponding to the standard front dip of 10 mm is the same, while the touch weight is adjusted to a similar value for both types of digital. From the purely mechanical standpoint, therefore, it should be immaterial whether the hammer is operated from a black or a white digital.

As some tuners have the regrettable habit of taking the A_3 as they find it, instead of first setting its frequency to that of the standard fork, it is not uncommon to meet with pianofortes which have, in course of time, dropped a semitone, or more, in pitch. Suppose such an instrument to have sunk precisely a semitone below its nominal pitch. Then playing upon it in the key of A ought to be precisely equivalent to playing upon a similar instrument, whose pitch has been maintained at its proper level, in the key of A flat. Under these circumstances a performer, playing the same composition on the respective instruments in the respective keys of A and A flat should actually be playing notes of the same absolute pitch, though employing different digitals for the purpose in the two cases. If this is so, and differences of key effect are still discernible, they must, presumably, be attributed to the different relative number and distribution of the black and white digitals involved. To settle the point it would not be difficult to tune one pianoforte exactly a semitone out with respect to another. Thus having tuned pianoforte I in the customary manner, the A on pianoforte II would then be tuned to the A flat on pianoforte I, the B flat on pianoforte II to the A on pianoforte I, and so on throughout. When, subsequently, a composition is played, say, in A flat on pianoforte I and in A on pianoforte II the absolute pitch of the corresponding notes will be identical, though the digitals employed for the purpose will not correspond. An investigation along these lines should serve to discriminate between the theories in question. To obviate the human element, which may enter subconsciously in the execution of a piece in two different keys, suitable rolls could be cut for use on player pianofortes.

Finally, there is no reason why a carefully calibrated set of forks should not be used to obtain what would almost certainly prove to be a closer approximation to equal temperament than is normally secured on a pianoforte, and then to ascertain the opinion of a company of musicians upon the effect of transposition.

What is needed to clear up these debatable points is experimental data produced by the joint efforts of the physicist, the expert tuner and the musician. It is, however, essential that the former should have at his disposal the resources of a properly equipped acoustical laboratory, and that both the latter should have had sufficient grounding in acoustics to grasp the points at issue. Otherwise no amount of willingness is likely to render their cooperation effective.

Sound Analysis

HARRY H. HALL
Cruft Laboratory, Harvard University, Cambridge, Massachusetts
(Received February 13, 1937)

Instruments for sound analysis may be grouped into five classes, graphic, resonance, heterodyne, stroboscopic, and diffraction analyzers. The operation of each type of instrument is briefly described. From the point of view of analysis sounds may be classified roughly into four groups: (1) Steady state sounds or sounds which may be maintained at constant fundamental frequency, constant intensity and unvarying quality for long enough to carry out the analysis; (2) Sounds which are essentially transient in nature; (3) Sounds which may be maintained constant on the average but whose intensity, frequency, and wave form are modulated at a constant frequency; (4) Noise, or sounds which are entirely random in form but which are continuously maintained. The sound spectra corresponding to each group are described and the application of the various types of instruments to the analysis of these spectra is discussed. It is pointed out that a graphic analyzer may not be used to analyze single waves taken from a short transient sound.

INTRODUCTION

ANY wave may be represented as the resultant of an array of sine waves having amplitudes, frequencies, and relative phases determined by the original wave. If this is a periodic disturbance of constant form or steady state wave, the component sine waves have frequencies which are integral multiples of a fundamental frequency and form a line spectrum. But if the original wave is a nonrecurrent pulse, the components have all frequencies and are distributed throughout a continuous spectrum. Sound analysis may be defined as the detection and resolution of such spectra formed by sound waves.

It follows that an ideal analysis must furnish,

(1) The frequencies of all components;
(2) The relative amplitudes of the components;
(3) The relative phases of the components at some given instant.

If this data is given the original wave may be reproduced precisely. But the human ear is not usually sensitive to phase and thus for many purposes a specification of the frequencies and amplitudes is often sufficient.

METHODS OF ANALYSIS

If the form of a given wave may be expressed as an analytical function or is available as a plot of displacement with time the analysis may be obtained by direct computation. This is a laborious process which may be avoided under proper conditions by instrumental methods. Instruments designed to accomplish this may be classed as follows:[1]

(1) Graphic analyzers
(2) Resonance analyzers
(3) Heterodyne analyzers
(4) Stroboscopic analyzers
(5) Diffraction analyzers

GRAPHIC ANALYZERS

The graphic analyzer requires that the wave be recorded as a plot of displacement with time. This is enlarged to proper dimensions and placed on a horizontal surface over which the analyzer may travel. One form of this instrument, the Henrici analyzer, shown in Fig. 1, has been described by Miller.[2] The stylus of the analyzer is made to trace the curve and the frequencies and amplitudes of the components may be read on the proper dials. Amplitudes of sine and cosine series are given which are equivalent to specifying the amplitudes and phases of a single series. In this way the first five components of the wave are given. Retracing the wave with a different setting of the instrument furnishes the next five and so on up to thirty components. The method assumes that the wave analyzed is taken from a train of steady state waves and therefore gives the equivalent line spectrum.

[1] A more detailed description of the five types of analyzers was given in the following: H. H. Hall, J. Soc. Mot. Pict. Eng. **27**, 396 (1936).
[2] D. C. Miller *The Science of Musical Sounds* (Macmillan, 1926), Lecture IV, p. 92.

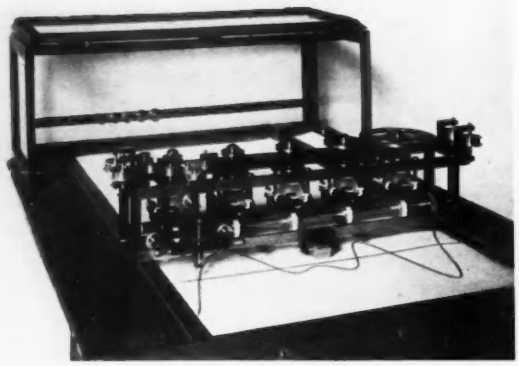


FIG. 1. The Henrici analyzer. After the index stylus has been made to trace the curve the amplitudes of the first five sine and cosine components may be read from the proper dials. A second tracing furnishes the next five and so on up to thirty components. Courtesy of D. C. Miller.

RESONANCE ANALYZERS

Suppose the frequency range which it is desired to analyze be divided by small steps into a large number of frequencies, and that an array of vibrating reeds be provided, one tuned to resonance at each frequency. If a complex steady state wave is applied to this assembly, those reeds whose frequencies are nearest to components in the wave will be set into motion and their amplitudes will depend on the amplitudes of the components. Such an instrument has been described by Hickman[3] and may be called a resonance analyzer. It will furnish the frequencies and amplitudes of the components but not their relative phases. Other resonators may replace the reeds such as electrical tuned circuits, tuned air chambers,[4] or band pass filters. Fig. 2 represents an instrument of the last type described by Freystedt.[5] There are in all twenty-two band pass filters, three per octave, covering the range from 40 to 5500 cycles. The outputs of the filters are connected one after another through a switching mechanism to the vertically deflecting plates of a cathode ray tube. As the tube is switched from one output to another the spot is moved a short distance horizontally so that the pattern traced on the tube is a series of vertical lines whose heights are proportional to the outputs of the filters and whose positions correspond to their frequencies. This form of analyzer is not subject to the arbitrary assumptions which apply to the graphic analyzer since it responds to whatever frequencies are actually present in the wave and does not assume that the wave is in a steady state.

The resolving power of such an instrument depends inversely upon the damping of the individual selecting elements. Likewise the length of time required for an analysis is inversely proportional to the damping and therefore directly to the resolving power.

HETERODYNE ANALYZERS

The multiplicity of resonators of the resonance analyzer is avoided in the heterodyne[6] or search tone analyzer. Referring to Fig. 3 the wave to be analyzed and a wave whose frequency can be varied are applied to a detector. The output will now contain new components of which some will have frequencies equal to the differences between those of the components of the applied wave and that of the variable wave. When the frequency of the latter is varied, the frequencies of this group of components will vary with it. In this way and with but one resonator or filter responding to a narrow band of frequencies, the frequencies of the components of the difference frequency band may be varied until, one by one, they are passed by the filter and actuate some indicating or recording device. It may be shown that the amplitudes of these components are proportional to those in the original wave.

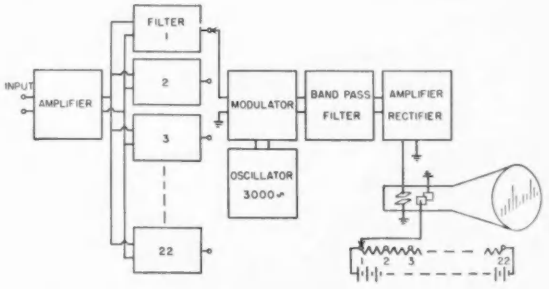


FIG. 2. Functional diagram of Freystedt's resonance analyzer. The switch connecting the filters to the measuring circuit and that connecting the horizontally deflecting plates of the cathode-ray tube to the battery are rotary and are mounted on the same shaft. The oscillator merely changes the whole frequency band by heterodyning to a more convenient range.[5]

[3] C. N. Hickman, J. Acous. Soc. Am. **6**, 108 (1934).
[4] C. W. Hewlett, Phys. Rev. **35**, 359 (1912).
[5] E. Freystedt, Zeits. f. tech. Physik **16**, 533 (1935).
[6] O. H. Schuck, Proc. I. R. E. **22**, 1295 (1934); also H. H. Hall, J. Acous. Soc. Am. **7**, 102 (1935).

It is at once clear that an analysis cannot be made very rapidly by this method because the resonator requires appreciable time to reach full amplitude of vibration after the wave has been applied, and this interval must elapse as many times as there are components to be measured. These analyses therefore require a longer time than those performed by the resonance analyzer since the various resonators in the latter are excited simultaneously. But it has the advantage of simpler construction and of being capable of indicating any frequency in a continuous range instead of in a range divided into finite steps. The time required for the analysis and therefore the time during which the wave may not depart appreciably from a steady state may be shown to be proportional to the square of the resolving power of the instrument. Nevertheless quite rapid analyses may be made with the heterodyne analyzer. The instrument built by the author will analyze the range from 50 to 10,000 cycles in less than four seconds and yet resolve components only 50 cycles apart if they are nearly of the same amplitude.

Stroboscopic Analyzers

A form of stroboscopic analyzer has been described by de Nemes[7] in which a rotating disk is used for wave analysis. The face of a disk is divided into concentric rings and each ring is divided into segments alternately black and white. The second ring from the center of the disk has twice as many segments as the first ring, the third has three times as many and so on. The disk is now rotated at constant speed fast enough to cause the segments of the first ring to pass a given point with a frequency equal to the fundamental frequency of the wave to be analyzed. If the disk is illuminated with light whose intensity is varied to correspond to this wave, the rings corresponding to components in the wave will appear to stand still. This assumes of course that the wave is in a steady state and that only component frequencies are present which are integral multiples of the fundamental frequency. The instrument indicates phases very nicely but the intensities of the components are not clearly given.

[7] T. de Nemes, Phil. Mag. **18**, 303 (1934), also Arch. f. Elektrotechnik, **26**, 403 (1932).

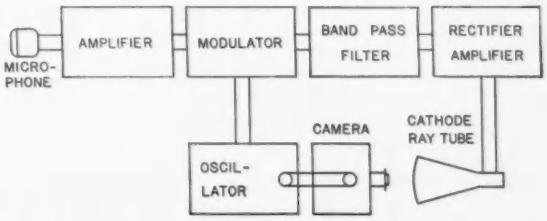


Fig. 3. Functional diagram of a heterodyne analyzer.

Diffraction Analyzers

There are two types of diffraction analyzer, one of which employs diffracted sound and the other diffracted light. An example of the first type described by Meyer[8] makes use of a diffraction grating designed for sound waves. Since the wave-length of audible sound may be several meters, and since the grating space must be comparable in width to one wave-length, it is necessary to change the wave-lengths to more convenient values. This is done by heterodyning to 45 kilocycles with a fixed frequency oscillator. The sound is radiated by a small ribbon loud speaker and falls upon a concave grating made of steel rods, which has a theoretical resolving power of 125 cycles and a dispersion of 8 cm per kilocycle. The spectrum is explored by means of a small condenser microphone which controls an oscillograph.

The advantage of this method lies in the very short time for the spectrum to be formed. This interval is the difference between the times required by the sound to travel from the nearest and farthest grating apertures to the microphone. Meyer gives one one-hundredth second for this time. Theoretically at least, it would be possible to observe the spectrum of a transient sound, which should appear continuous. But this is not possible at present because there is no known "photographic plate" for sound.

Another ingenious diffraction analyzer makes use of diffracted light to produce the spectrum. Observation of the spectrum produced by a diffraction grating may yield facts concerning the form of the grating. Thus if the spectrum formed by monochromatic light contains in addition to the single bright line a number of "ghosts," the imperfections of the grating may be computed from their positions and intensities.

[8] E. Meyer, J. Acous. Soc. Am. **7**, 88 (1935).

These ghosts are the Fourier series terms which represent the distribution of the widths of the diffracting elements with their positions along the grating. Thus if these widths were determined by the wave to be analyzed a spectrum of ghosts would be formed with monochromatic light which would be an analysis of the wave. This has been accomplished by Germansky[9] in the following manner. A polished metal surface upon which the grating is to be made is covered with a thin coating of a mixture of glue, ammonium bichromate and chromic acid, which has the property of becoming insoluble in water when exposed to light. A variable density photograph of the wave to be analyzed is prepared and the treated metal surface is exposed to light through the variable density reproduction of the wave and a fine screen of alternately transparent and opaque parallel lines perpendicular to the axis of travel of the wave. After exposure the unaffected coating of glue and bichromate is washed off leaving a grating whose lines vary in width in the proper way. The method is similar to that by which ordinary halftone plates are made except that a screen of parallel lines is used in place of a fine mesh. Such an analyzer has the advantage that the spectrum may be recorded directly upon a photographic plate. In the state in which it was described the method cannot compare with the sound diffraction analyzer, yet if there were some means of producing the grating simultaneously and continuously with the sound, the scheme would have a number of advantages.

Discussion of Sound Spectra

From the point of view of analysis, sounds may be classified roughly into four groups:

(1) Steady state sounds or sounds which may be maintained at constant frequency, constant intensity and unvarying quality for long enough to carry out the analysis;
(2) Sounds which are essentially transient in nature;
(3) Sounds which may be maintained constant on the average but whose frequency, intensity and wave form are modulated at a constant frequency;
(4) Noise, or sounds which are random in form but which are continuously maintained.

[9] B. Germansky, Ann. d. Physik **7**, 453 (1930).

Steady State Sounds

Steady state sounds represent an array of components whose frequencies are integral multiples of the fundamental frequency, and whose amplitudes are determined by the character of the sound. These sounds may be analyzed generally by all the types of analyzers which have been described. The analyzer must have sufficient resolving power to separate the various components of the wave. If it has not, the indication in a particular frequency region will be the resultant of all components embraced by the selecting element of the analyzer, which is a fluctuating quantity. The result is an incomplete analysis of the wave.

Transient Sounds

Transient sounds represent continuous spectra whose amplitude and phase distributions are determined by the sound. These spectra are considered to exist throughout time, but may be observed only during the lifetime of the original sound and are not fully indicated until the full sound has passed. In order to detect these spectra an analyzer must be capable of indicating the presence of all frequencies throughout the duration of the sound. Fig. 4 shows the well-known amplitude distribution of the spectrum computed for the pulse of sinusoidal oscillation at the top of the figure. Short syllables such as "tip" or "top" are more or less of this character. If spoken by a man these words contain some twelve to fifteen individual waves at a frequency between 100 and 150 cycles exclusive of the transitional oscillations at the beginning and end. The pulse of Fig. 4 contains fifteen waves at a frequency of 125 cycles and is intended to be comparable with such sounds. The width of the central maximum depends upon the length of the pulse, its width in cycles being equal to twice the reciprocal of the duration of the pulse in seconds. If the length of the pulse is increased the side bands are reduced with respect to the central maximum and the whole pattern contracts until in the limiting case of a steady state wave, a single component at the frequency of the generating wave is left. In the case shown in Fig. 4 the secondary minima nearest to the central maximum are about thirteen decibels below the

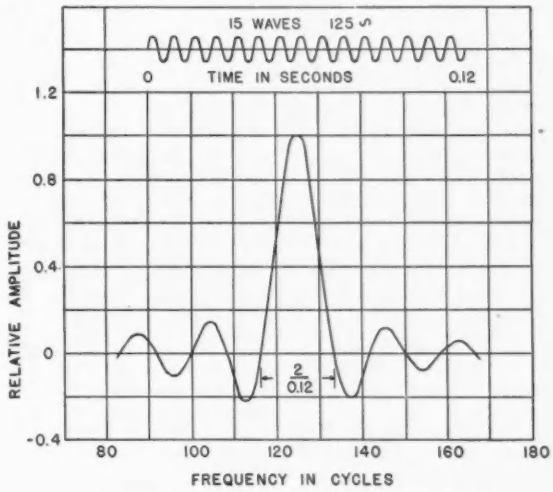


FIG. 4. Frequency spectrum computed for the pulse of sinusoidal oscillation shown at the top of the figure.

latter in amplitude, and displaced from it by 12.5 cycles on either side. This is about three quarter tones. The fact that this part of the curve is negative means that the components under it are of opposite phase with respect to those of the central maximum. The next pair of maxima are about seventeen decibels below the central maximum, and are displaced by about 21 cycles on either side of it—about one and a quarter tones. According to masking data these secondary bands of frequencies are probably audible and should affect the quality of the sound. Of course the word "tip" contains complex waves rather than simple ones and its spectrum will be more complex than that of Fig. 4. It may be shown that for each component that a complex wave would have if it were repeated in a steady state there will be an individual pulse spectrum such as that of Fig. 4, and that the amplitude at any frequency will be the resultant of the components at that frequency of all of the pulse spectra.

Only diffraction analyzers are capable of making analyses of this type of sound, and even with these instruments in their present state it is not possible to obtain a complete resolution of the spectrum. If a Henrici analyzer is used to analyze selected waves taken from different parts of such a short sound, it will not give the true spectrum, but will give the line spectrum which would be produced if the particular wave in question were being repeated in a steady state. This will be very different from the true spectrum if the sound is of short duration. If the sound lasts for a fairly long time and changes but slowly in wave form the discrepancies between such analyses and the true spectra are not as serious because the pulse spectrum approaches the line spectrum in form as the duration of the pulse is increased. It is difficult to state a criterion by which a "long" pulse may be distinguished from a "short" one. This will depend upon the purpose for which the analysis is made. If the measurements concern the effect of these sounds upon the ear, then a long pulse may be defined as one which at a given instant within the pulse sounds like a steady state wave of the same form. This means that the pulse must have been long enough to reduce the secondary maxima below the masked threshold of hearing.

Within these restrictions the resonance and stroboscopic types of analyzers may also be used.

MODULATED SOUNDS

Sounds which are modulated at a constant frequency, are strictly steady state sounds and may be analyzed by any analyzer having sufficient resolving power. Here again if the modulating frequency is so low that the components produced by the modulation are below the masked auditory threshold it is permissible to make use of a Henrici analyzer to analyze selected waves taken from different portions of the modulation cycle. The result will be a number of line spectra each representing what would be heard if the particular wave in question were being repeated

TABLE I.

TYPE	INPUT	TIME FOR ANALYSIS	WAVES ANALYZED	RESULTS
Graphic	Oscillogram	1 hour	Steady State Slowly varying Transient	Frequency Amplitude Phase
Resonance	Microphone	1/10 second Proportional to Resolving Power	Steady State Slowly varying Transient	Frequency Amplitude
Heterodyne	Microphone	4 seconds Proportional to Square of Resolving Power	Steady State	Frequency Amplitude
Stroboscopic	Microphone	1/20 second	Steady State Slowly varying Transient	Frequency Phase
Diffraction	Microphone	1/100 second	Steady State Transient	Frequency Amplitude

in a steady state. In the case of vowels sung with vibrato it is doubtful whether this is true. Kock[10] reports that a frequency vibrato of small amplitude cannot be distinguished by the ear from an amplitude vibrato although the listener is aware that modulation of some kind is present. Thus the ear does not appear to follow the frequency or amplitude fluctuations as such. It is more likely that the ear partially analyzes such a sound into varying components. If the analysis were complete a perfectly steady state spectrum would result and no impression of modulation would be received. This suggests the possibility of determining the resolving power of the ear—by finding the lowest modulation frequency which the ear fails to detect as modulation.

[10] W. E. Kock, J. Acous. Soc. Am. **7**, 56 (1935).

Noise

Random oscillations of this type represent continuous spectra which are constant on the average. They may be analyzed by all types of analyzers with the exception of the graphic and stroboscopic types. The result will be the average distribution of amplitude with frequency for these sounds.

Table I gives a classification of various types of analyzers.

Current Publications on Acoustics

F. A. FIRESTONE
147 East Physics Building, University of Michigan, Ann Arbor, Michigan

Reviews of Contemporary Papers

MEMBERS who have occasion to translate important papers from foreign journals will be doing a real service to those members who do not read the foreign languages easily, if they will prepare a review of such articles for publication in this section.

These reviews should be of the nature of a lengthy abstract (500 to 1500 words) rather than a critique or appraisal, and should attempt to set forth in this limited space as much of the original author's contribution as possible. One or two figures may be included if desired.

To avoid duplication of effort it is desirable to arrange with the office of this department before writing a review.

Professor Erwin Meyer and one of his associates, Mr. Schudrzyk, have kindly agreed to supply us with a number of reviews of important German papers for each issue.

Recent Problems in Supersonics. L. BERGMANN. Akustische Zeits. **3**, 118 (1936).—In this paper a short survey is given of the recent developments in supersonics. Up to 60,000 cycles/sec. (120,000 cycles/sec. exciting harmonics) magnetostrictive methods are used for the production of supersonic vibrations. Such transmitters, however, have a large temperature coefficient. For higher frequencies, only piezoelectric means are of any practical value. Wave-length and consequently velocity of propagation are determined either with Pierce's interferometer or by optical methods. The index of refraction, depending on the condensation of the medium, varies periodically along the direction of propagation. This gives rise to the well-known diffraction phenomena, the distance between the striae being proportional to the wave-length. Special attention is to be called to one direct procedure for determining the velocity of propagation which is somewhat similar to Foucault's measurement of the velocity of light. A stationary picture of the progressive supersonic wave is obtained by a "Schlieren" arrangement and a rotating mirror. The angular velocity of the mirror which renders the picture stationary is proportional to the velocity of sound.

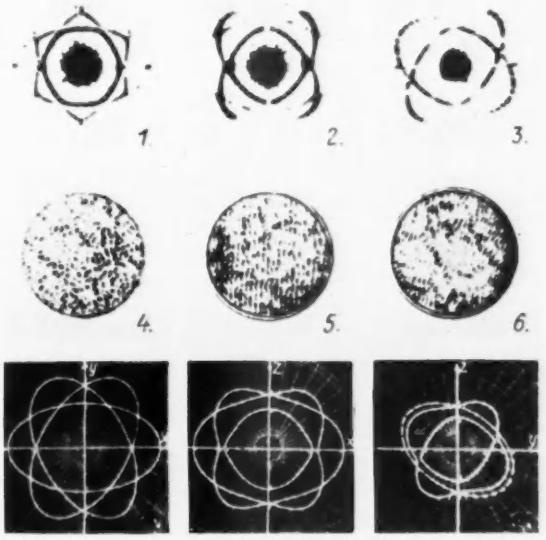


FIG. 1. Optical diffraction diagrams for a quartz crystal.

From supersonic measurements, constants of the materials can be determined as: the adiabatic modulus of elasticity, the ratio of the specific heats, from the variation of the compressibility with concentration, the valence of the ions, etc. Also the constitution of some organic and inorganic liquids could be deduced. Quite recently, absorption measurements have become a means of investigating intermolecular processes. There are many technical applications of supersonics. Fractures in materials can be ascertained. Mixtures as water-oil, or water-mercury are transformed into quite stable emulsions. During electrolysis, exposing the cathode to intense supersonic radiation very fine distributions are given off. Photographic emulsions treated with supersonic sound are more stable and more homogeneous. Liquids are degasified, the bubbles travel to the nodes where they unite, and ascend to the surface.

One remarkable application of supersonics lies in the determination of the three elastic constants of a transparent body. When excited in one of its higher modes of vibration, waves traverse it in all directions. At the nodes condensation, at the troughs, rarefaction of the material takes place. The optical grating thus formed gives rise to peculiar diffraction phenomena quite similar to the well-known Laue diagrams in x-rays. Knowing the elastic constants of the material, such diagrams can be calculated by means of a theory given by Fuess and Ludolf. Conversely, the elastic constants can be deduced to a very great degree of accuracy from such diffraction pictures. If the body is opaque, reflection on its surface leads to similar results. In parts 1–3 of Fig. 1, diffraction diagrams are given as obtained for a quartz crystal when transluced in the direction of the optical axis (Z), the X and the Y (polar) directions, respectively. Parts 4–6, obtained by "Toepler's Schlieren method," illustrate the intricate vibrations of such a crystal, and finally, parts 7–9 show the calculated results.

E. J. SKUDRZYK

The Onset of Sound on an Organ. F. TRENDELENBURG AND E. THIENHAUS. Akustische Zeits. **2**, 59 (1936).—The authors have performed a series of investigations on the transients of organ pipes, in the Eosander Chapel in Berlin, on an organ which is a musically very valuable specimen (constructed in 1706).

The microphone was placed close to the pipes and the reverberation time of the chapel was very small (less than 1 sec.) so that its acoustics did not interfere. The amplified microphone output was divided into two parts, one going directly to an oscillograph, while the other was passed through an octave filter (octave band pass frequency analyzer) to another oscillograph loop. In this way the total sound could be compared with its frequency components. Records were taken of the most characteristic pipes as the principal 8′ (open cylindrical pipe, normal scale), the Lieblich Gedackt 8′ (a very soft sounding closed prismatic wooden pipe of a small scale), and some reed pipes, the vox humana, trumpet and the trombone. In the original paper, the results are tabulated in detail and are followed by a series of oscillograms.

There is found to be a fundamental difference between the onset of sound in a flue pipe and a reed pipe. The building up of sound in a flue pipe is very slow. The fundamental of the principal 8′ (C) attained 0.8 of its final value only after 0.5 sec., this time being almost 0.2 sec. for the c^0 and 0.1 sec. for the c^1 and c^2. Reed pipes have comparatively short transients. In case of the trumpet 8′, the stationary state was reached after about 0.1 sec., i.e., 5 times more rapidly than in the corresponding previous case. Higher pitches build up quicker. In flue pipes, the expressed transients seem to be due to a periodic sequence of vortices at the mouth of the pipe which only build up slowly. Reed pipes always build up in the same way, but flue pipes vary in certain details, probably because the action of the mechanical slide depends on the way the key is depressed.

For the Lieblich Gedackt, the tone starts with a sinusoidal oscillation which is not an integral multiple of the fundamental, but about $5\frac{1}{2}$ times its frequency, lasting for 0.25 sec. In the steady state this oscillation has died away leaving only the fundamental and a faint third harmonic. The Lieblich Gedackt is worked at a blast pressure only slightly greater than the minimum operating pressure, resulting in: 1st, a slow rise of pressure at the mouth (operating pressure being obtained after 0.2 sec.); 2nd, an abrupt excitation of different partial tones not lying harmonically to the fundamental, the latter appearing at last. The partial tone at the onset of the Lieblich Gedackt seems to be particularly favored on account of the construction and the scale of this set of pipes. For the intonation of the Lieblich Gedackt, the characteristic transients are of uppermost importance. Subjectively, they are quite noticeable and may even predominate, when rapid sequences are played. Similar considerations apply to the principal 8′, where a sudden onset of the overtones almost causes the impression of a beat. The reed pipes as trumpet and trombone show at their onset only one frequency component noticeably displaced from the steady state value of the fundamental, and this seems to be due to an excitation of the air resonance, when the shallot is still open.

Very peculiar are the transients in the vox humana, where the building up of sound is somewhat slower than in the other reed pipes. The oscillations start with a low frequency component and apparently when the tongue has gained sufficient amplitude and strikes against the shallot the higher frequencies appear. At this instant, the stationary state is established, as, owing to the shallot, the amplitudes of the tongue cannot vary any more. The stationary sound of the vox humana is also very interesting; the higher octaves are seen to be excited and decaying periodically, in sudden kinks, and this is the reason for the well-known similarity of the vox humana to the human voice.

The principal differences in the building up of sound in the various reed pipes are caused by the different construction of the tongues, trumpet and trombone having short, broad and strong tongues, while that of the vox humana is long and thin. It has to be noted that the low frequency transient at the onset of a pipe has not the frequency of the fundamental. In case of the c, c^0, c^1 and c^2 the low frequency transients were: 45, 141, 160 c/sec., respectively, corresponding to the fundamental frequencies: 61, 121, 242, 480, c/sec. This might be a coupling phenomenon. The vibrational modes of the resonators of the pipes are usually much higher than the fundamental frequencies.

E. J. SKUDRZYK

Three New Journals

Three new journals have come to the attention of the Editor, and since these publications deal more or less with acoustics, they give further evidence of the increase of activity in the subject. We welcome these important new magazines.

The *Akustische Zeitschrift* appeared in September under the direction of Dr. Martin Grützmacher and Dr. Erwin Meyer. This journal plans to publish information concerning scientific and technical investigations in the different domains of acoustics. It is likely to become the medium for publications that otherwise might appear in the *Electrische Nachrichten-Technik* (usually abbreviated "E.N.T.") or in the *Zeitschrift für Hochfrequenztechnik*. It is the intention to publish at least six numbers of the magazine each year The cost is 10 marks yearly, and the printer is S. Hirzel in Leipsic.

A second new publication, closely associated with acoustics, is the *RCA Review*, a quarterly journal of radio progress, published by the RCA Institutes Press, New York City. As its name indicates, it will publish articles, many of which will interest members of the Acoustical Society.

The third new venture is the *Journal of Applied Physics*, published by the American Institute of Physics; it is "devoted to physics in its role as the science basic to other natural sciences and to the arts and industries. Previous to 1937 this publication was known as *Physics*. It aims to be of service not only in physical laboratories, but also in the laboratories of industrial, chemical, geological, geophysical, meteorological, radio, and similar concerns. *The Journal of Applied Physics* publishes editorials, news of physicists and reviews, as well as technical papers of applied physics." The subscription price is $5.00 per year to members of the Acoustical Society and $7.00 per year to nonmembers of societies affiliated with the Institute. Dr. Elmer Hutchisson, assistant director of the Institute, is Editor of this journal.

F. R. WATSON

References to Contemporary Papers on Acoustics

MOST of the titles of papers appearing in other languages than English have been translated; the clue as to the language in which the article is written can usually be found by noting the name of the journal in which it appears. Abstracts in English of many foreign papers have appeared or soon will appear in Science Abstracts, Section A. The abbreviations of the names of journals are those used in Science Abstracts and can be found in any annual index to those abstracts.

General, Unclassified

Sound Absorption of Gases in Kundt's Tubes, Especially at Low Pressure. Herman Oberst. Zeits. f. techn. Physik **17**, 580–582 (1936) Nr. 12.

On the Early History of Acoustics. Hans Schimank. Zeits. f. techn. Physik **17**, 500–508 (1936) Nr. 12.

The Importance of Acoustics in the Whole Realm of Physics and Technology. Erwin Meyer and E. Waetzmann. Zeits. f. techn. Physik **17**, 508–512 (1936) Nr. 12.

The Electrical Transmission of Speech and Music. Hans Joachim v. Braunmuhl. Zeits f. techn. Physik **17**, 539–549, (1936) Nr. 12.

Die Akustik auf dem Physikertag in Bad Salzbrunn. F. Trendelenburg. Naturwiss. **24**, 809 (1936).

Theory of Waves and Vibrations

An Asymptotic Condition in the Forced Vibration of Plates at High Frequency. A. Schoch. Zeits. f. techn. Physik **17**, 582–584 (1936) Nr. 12.

The Design of a Mechanical Analogy for the General Linear Electrical Network with Lumped Parameters. Myron Pawley. J. Frank. Inst. **223**, 179 (1937).

The Vortex Structure in Edge Tones. F. Krüger and H. Casper. Phys. Zeits. **37**, 842–849 (1936). Also in Zeits. f. techn. Physik **70**, 416–423 (1936) Nr. 11.

The Small Vibrations of a Pendulum with Oscillating Support. Artur Erdelyi. Zeits. f. angew. Math. u. Mech. **16**, 171–182 (1936) Nr. 3.

Interference Between Waves Having a Modulated Frequency. J. Brillouin. R.G.E. **40**, 434–441 (1936). Sci. Abs. **4**, 5624 (1936).

Some Special Problems of the Forced Torsional Vibrations of an Elastic Circular Cylinder of Finite Length. Yukiti Nomurs. Sc. Rep. Tohoku Univ. **25**, 11 (1936). In English.

Certain Vibration Problems Solved by Means of an Analogous Problem in Heat Conduction. George Green. Phil. Mag. **22**, 1079 (1936).

Sound Sources

On the Equation of Vibrations of a Plate. A. Weinstein. Soc. de phys. de Geneve **53**, 62–64 (1936) Nr. 2.

On the Free Vibration of a Circular Plate, Clamped at the Edge with Friction. Georg Ostroumow. Techn. Phys., U.S.S.R. **3**, 583–603 (1936) Nr. 7. In German.

Contribution to the Theoretical and Experimental Investigation of the Flexural Vibration of Rectangular Plates with Free Edges. Bohuslav Pavlik. Ann. d. Physik **27**, 532 (1936).

Supersonic Vibrations and Waves

On the Visibility of Ultrasonic Waves. P. Rama Pisharoty. Proc. Indian Acad. **A4**, 27–36 (1936), Nr. 1.

On a Very Simple Method of Detecting Supersonic Waves in Liquids. Z. Bergmann and H. J. Gochlich. Phys. Zeits. **38**, 9–13 (1937) Nr. 1.

Action of Acoustic and Supersonic Vibrations on Aerosols. E. Heidmann. Kolloid Zeits. **77**, 168–172 (1936). Sci. Abs. 303 (1937).

Diffraction of Light by Supersonic Waves. S. M. Rytov. Comptes rendus de l'Acad. des Sience, U.S.S.R. **3, 4,** 197–156 (1936).

Absorption of Supersonic Waves in Acetic Acid. P. Bazulin. Sci. Abs. 305 (1937).

Optical Researches on Supersonic Fields in Liquids and Glasses. E. Heidmann and K. H. Heesch. Zeits. f. Physik **104**, 197–206 (1937).

Formation of Fogs by Supersonic Waves. K. Sollner. Trans. Faraday Soc. **32**, 1532–1536 (1936). Sci. Abs. A5632 (1936).

Visible Cavitation Caused by Supersonic Waves. K. Sollner. Trans. Faraday Soc. **32**, 1537–1539 (1936). Sci. Abs. A5634 (1936).

Action of Supersonic Waves in Suspensions. F. J. Burger and K. Sollner. Trans. Faraday Soc. **32**, 1598–1603 (1936). Sci. Abs. A5634 (1936).

New Problems in the Field of Supersonics. L. Bergmann. Zeits f. techn. Physik **17**, 512–518 (1936) Nr. 12.

Supersonic Surface Waves and Their Optical Detection. H. Ludloff. Zeits. f. techn. Physik **17**, 522-528 (1936) Nr. 12.

Supersonic Velocities in Organic Liquids. Part III. Esters and Ethers. S. Parthasarathy. Proc. Indian Acad. **3**, 519–522 (1936) Nr. 6.

Diffraction of Light by Supersonic Waves. Proc. Indian Acad. **3**, 442–447 (1936) Nr. 5. Also Part II, **3**, 594–606 (1936) Nr. 6.

The Diffraction of Light by High Frequency Sound Waves. Part V. General Considerations. Oblique Incidence and Amplitude Changes. Proc. Indian Acad. **3**, 459–465 (1936) Nr. 5.

Theory of Sound Dispersion. L. Landau and E. Teller. Phys. Zeits. d. Sowjetunion **10, 1,** 34–43 (1936). Sci. Abs. A5214 (1936).

Diffraction of Light by Supersonic Waves: Generalized Theory. N. S. N. Nath. Indian Acad. Sci. Proc. **4A**, 222–242 (1936). Sci. Abs. A5219 (1936).

Method of Making Stationary Supersonic Waves in Liquids Visible. D. Nomoto. Proc. Phys. Math. Soc. Japan, **18**, 402–424 (1936). Sci. Abs. A5221 (1936).

Light Modulation by Supersonic Waves. H. E. R. BECKER. Hochfrequenztechn. u. Elektroakustik. **48**, 89–91 (1936). Sci. Abs. A5222 (1936).

Resonance Curves for a Quartz Oscillator Immersed in Liquids. Proc. Indian. Acad. **A3**, 544–553 (1936) Nr. 6.

The Dispersion of Sound in Oxygen. LESTER S. SINNESS AND W. E. ROSEVEARE. J. Chem. Phys. **4**, 427–431 (1936) Nr. 7.

The Aggregation of Suspended Particles in Gases by Sonic and Supersonic Waves. O. BRANDT AND E. HIEDEMANN. Trans. Faraday Soc. **32**, 1101–1110 (1936) Nr. 8.

The Coagulation of Smoke by Supersonic Vibrations. E. N. DA C. ANDRADE. Trans. Faraday Soc. **32**, 1111–1115 (1936) Nr. 8.

Experiments on Coagulation by Supersonic Vibrations. R. C. PARKER. Trans. Faraday Soc. **32**, 1115–1119 (1936) Nr. 8.

An Effect of Supersonic Radiation on Electrodeposits. WILLIAM T. YOUNG AND H. KERSTEN. J. Chem. Phys. **4**, 426–427 (1937) Nr. 7.

Visibility of Ultrasonic Waves and Its Periodic Variations. N. S. N. NATH. Proc. Indian Acad. Sci. **4A**, 262–274 (1936). Sci. Abs. A5625 (1936).

Theory of Acoustic Coagulation. O. BRANDT, H. FREUND AND E. HIEDEMANN. Kolloid Zeits. **77**, 103–115 (1936). Sci. Abs. A5631 (1936).

Sound Dispersion in Liquids. E. HIEDEMANN, N. SEIFEN AND E. SCHREUER. Naturwiss. **24**, 681 (1936). Short note.

Ultrasonic Velocities in Liquid Mixtures. ERNST LUBEKE, S. PARTHASARATHY. Proc. Indian Acad. **3**, 297 (1936).

On the Influence of Supersonic Waves on Chemical Reactions. S. SOKOLOW. Techn. Phys. U.S.S.R. **3**, 176 (1936). In French.

Photometric Study of the Refraction of Light by Supersonic Waves in Liquids and Gases. WILHELM KORFF. Phys. Zeits. **37**, 708–720 (1936).

Theory of Light Refraction by Supersonic Waves. E. DAVID. Verh. d. D. Phys. Ges. **17**, 8 (1936).

Stroboscopic Phenomena on Passing Light Through Two Supersonic Waves. R. BAR. Helv. Phys. Acta. **9**, 678 (1936).

ARCHITECTURAL ACOUSTICS

The Effect of Air Coupling in Acoustic Insulation by Elastic Supports. A. H. DAVIS AND A. E. KNOWLER. Phil. Mag. **23**, 154–157 (1937).

The Sound Insulation of Single and Complex Partitions. J. E. R. CONSTABLE AND G. H. ASTON. Phil. Mag. **23**, 161–180 (1937).

Resonances of Closed and Open Rooms, Streets and Squares. G. JÄGER. Akad. Wiss. Wien, Per. 145, 2a. **1–2**, 73–93 (1936). Sci. Abs. 306 (1937).

Room Acoustics Considered Geometrically. F. M. OSSWALD. Zeits. f. techn. Physik **17**, 561-563 (1936) Nr. 12.

The Physical Basis of Room Acoustics. L. CREMER. Zeits. f. techn. Physik **17**, 528–539 (1936) Nr. 12.

The Natural Frequencies of Closed and Open Rooms. GUSTAV JÄGER. S. A. Wiener Ber. **145**, 73–93 (1936).

Resonance Absorption of Sound. S. RSCHEVKIN. Techn. Phys., U.S.S.R. **3**, 560–576 (1936).

Resonance Effects in Rooms, Their Measurement and Simulation. W. BURCK, P. KOTOWSKI AND H. LICHTE. E. N. T. **13**, 268–279 (1936).

The Effect of an Acoustically Absorbent Lining Upon the Sound Insulating Value of a Double Partition. J. E. R. CONSTABLE. Proc. Phys. Soc. **48**, 690 (1936). Theory and experiment.

NOISE, SOUND MEASUREMENT

High Intensity Sound Waves. C. G. SUITS. Gen. Elec. Rev. **39**, 430–434 (1936). Sci. Abs. A5224 (1936). Motion pictures of spark sounds.

The Measurement of Acoustical Impedance by a Null Method. K. SCHUSTER. E.N.T. **13**, 164 (1936).

A Study of Characteristics of Noise. V. D. LANDON. Proc. I.R.E. **24**, 1514 (1936). A study of the peak and r.m.s. values of a random noise after passing through band pass devices of different band widths.

Noise Phenomena in Electrical Machinery. ERNST LÜBCKE. Siemens Zeits. **16**, 204 (1936).

ACOUSTICAL INSTRUMENTS AND APPARATUS

A Pointer Frequency Meter for Sound Wave Frequencies. Techn. Phys., U.S.S.R. **3**, 361–365 (1936).

The Measurement of Velocity with the Rayleigh Disc. MARTA CHYTILOVA. Publ. Fac. d. Sc. Univ. Masaryk. Number 226 (1936). In French. Comparison of circular and rectangular disks.

Combination Horn and Direct Radiator Loud Speaker. H. F. OLSON AND R. A. HACKLEY. I.R.E. **24**, 1557 (1936).

Measurement of Phases of Acoustical Pressures. Z. CARRIERE. Rev. d'Acoustique, **5**, 1–23 (1936). Sci. Abs. 309 (1937).

Hartmann Acoustic Generator. Engineering **142**, 491–492 (1936). Sci. Abs. 310 (1937).

Acoustical Measurements with the Sound Meter Connected to the Damping Recorder. H. G. THILO. Zeits. f. techn. Physik **17**, 558–561 (1936) Nr. 12.

The Influence of the Size and Shape of Baffles on the Sound Radiation of Loud Speakers. GERHARD BUCHMANN. Zeits. f. techn. Physik **17**, 563–568 (1936) Nr. 12.

Sound Wave Stroboscope. B. F. MCNAMEE. Electronics **9**, 24 (1936). The Foley method of spark sound photography repeated periodically to give a stroboscopic action.

SPEECH AND HEARING

Electrical Apparatus for Making Vowel Sounds. K. W. WAGNER. Preuss. Akad. Wiss. Berlin, Adhandlungen **2**, **44** (1936). Sci. Abs. 307 (1937).

Questions in the Limiting Region between Physical and Physiological Acoustics. F. TRENDELENBURG. Naturwiss. **25**, 49–59 (1937).

On the Psychophysics of Hearing. Monaural and Binaural Differential Sensitivity and Exposure Time. MORGAN UPTON AND ALFRED H. HOLWAY. Proc. Nat. Acad. Sci. **23**, 29–34 (1937).

Advances in the Physiology of Hearing. GEORG V. BEKESY. Zeits. f. techn. Physik **17**, 522–528 (1936) Nr. 12.

Speech Research with Filters and Oscillograph. FERDINAND TRENDELENBURG AND ERICH FRANZ. Wiss. Veroff. a. d. Siemens Werken **15**(2), 78 (1936).

Physical Nature of Certain of the Vibrating Elements of the Internal Ear. C. S. HALLPIKE, H. HARTRIDGE, A. F. RAWDON SMITH. Nature **138**, 839 (1936).

On the Nature of Aural Harmonics. S. S. STEVENS AND E. B. NEWMAN. Nat. Acad. Sci. Proc. **22**, 668 (1936).

On the Persistence of the Sensation of Speech. MARIO MARRO. Phil. Mag. **22**, 847 (1936). Experiments on the hearing of periodically interrupted speech sounds.

On Questions of Symmetry and Heredity in the Human Ear. E. WAETZMANN. Zeits. f. techn. Physik **17**, 549–553 (1936) Nr. 12.

On the Distribution of Speech Pressures in Carrier Frequency Transmission. D. THIERBACH AND H. JACOBY. Zeits. f. techn. Physik **17**, 553–557 (1936) Nr. 12.

Intelligibility and Loudness as a Function of Frequency and Amplitude Limits. F. STRECKER. Zeits. f. techn. Physik **17**, 568–572 (1936) Nr. 12.

Logarithmic and Linear Loudness Scales. W. BURCK, P. KOLOWSKI AND H. LICHTE. Ann. d. Physik **27**, 664 (1936).

Electronic Music and Instruments. B. F. MIESSNER. Proc. I.R.E. **24**, 1427 (1936). An excellent general survey of the subject.

On the Resonance Properties of Stringed Instruments. H. BACKHAUS. Zeits. f. techn. Physik **17**, 573–578 (1936) Nr. 12.

Research on the Beginning Vibrations of Organ Pipes. F. TRENDELENBURG. Zeits. f. techn. Physik **17**, 578–580 (1936) Nr. 12.

Acoustical Society News

THE next meeting of the Acoustical Society will be held in Washington, D. C., on Monday and Tuesday, May 3 and 4. It presents an opportunity for members not only to attend the Acoustical Society but also the American Physical Society, April 29–May 1, and the various interesting features of Washington. A program of the meeting will be mailed to members by Secretary Waterfall. Mr. V. L. Chrisler of the National Bureau of Standards is chairman of the committee on program and arrangements.

ATTENTION of readers is again called to the new department on Current Publications on Acoustics, the first installment appearing in this issue of the Journal. Professor F. A. Firestone, University of Michigan, Ann Arbor, Michigan, is in charge of this department.

FIFTH INTERNATIONAL CONGRESS FOR APPLIED MECHANICS (1938)

THE American Committee, to whom has been delegated responsibility for organizing the Fifth International Congress for Applied Mechanics by the International Committee at its meeting at Cambridge University, England, in July, 1934, announces that the Fifth Congress will meet in Cambridge, Massachusetts, U. S. A., September 12–16, 1938 at Harvard University and the Massachusetts Institute of Technology. As in the past, this congress is to be a meeting of persons working in the field of applied mechanics before whom reports of recent work may be presented for discussion.

The program will cover three main divisions of applied mechanics as follows:

I. Structures, elasticity, plasticity, fatigue, strength theory, crystal structure.
II. Hydro- and aerodynamics, gasdynamics, hydraulics, meteorology, water waves, heat transfer.
III. Dynamics of solids, vibration and sound, friction and lubrication, wear and seizure.

Following the meeting at Cambridge, Massachusetts, it is expected that arrangements will be made to visit Washington (National Bureau of Standards), and Langley Field (National Advisory Committee for Aeronautics).

Dormitory and boarding facilities will be made available by Harvard University.

Inquiries should be addressed to the Fifth International Congress for Applied Mechanics, Massachusetts Institute of Technology, Cambridge, Massachusetts.

INDEX TO VOLUME 8

A

ABBOTT, ERNEST J.: Role of Acoustical Measurements in Machinery Quieting, 133–142; (Abstract) 66
ACOUSTICAL SOCIETY NEWS, 64, 153, 204, 267
AMERICAN TENTATIVE STANDARDS FOR NOISE MEASUREMENT Z 24.2—1936, 143–146; for Sound Level Meters Z 24.3—1936, 147–152

B

BÁRÁNY, ERNST: Transposition of Speech Sounds, 217–219
BARNES, T. G.: (see Lindsay, R. B.) 42–47
BEDELL, E. H.: Some Data on a Room Designed for Free Field Measurements, 118–125; (Abstract) 69
BOOK REVIEWS
ELEKTROAKUSTISCHE UNTERSUCHUNGEN IN HALLRÄUMEN. Hans Frei, 64
THE NEW ACOUSTICS. N. W. McLachlan, 64
PSYCHOLOGY OF THE VIBRATO IN VOICE AND INSTRUMENT. Carl E. Seashore, 205
BRUDERLIN, HENRY H.: Developments in Aircraft Sound Control, 181–184
—— AND E. WHEATON: Developments in Aircraft Sound Control, (Abstract) 67

C

CHRISLER, V. L.: Variation of Sound Absorption with Area, (Abstract) 67
COLWELL, R. C. AND L. R. HILL: Mechanical Vibrations at Radiofrequencies, 60–61
COWAN, MILTON: (see Lewis, Don) 20–22
—— AND DON LEWIS: Influence of Intensity on the Pitch of Violin and 'Cello Tones, (Abstract) 66
CURRENT PUBLICATIONS ON ACOUSTICS
Reviews of Contemporary Papers, 263–264
References to Contemporary Papers on Acoustics, 265–267

D

DAVIS, H.: (see Stevens, S. S.) 1–13

E

EISENHOUR, B. E.: Control of Temperature Variation in the Frequency of Tuning Forks, (Abstract) 67

F

FIRESTONE, F. A.: (see Trimmer, J. D.) (Abstract) 207
—— Current Publications on Acoustics, 263–267
FREE, E. E. AND D. J. HAHN: Legal Problems of Noise, (Abstract) 66

G

GARDNER, M. B.: (see Steinberg, John C.) (Abstract) 207
GAULT, ROBERT H.: Studies of the Vibro-Tactile Senses as Means for Determining Direction, (Abstract) 65
GOODFELLOW, LOUIS D.: Rearranging Speech Spectra for Tactile Recognition, (Abstract) 65
GREENE, PAUL C.: Violin Intonation, (Abstract) 207

H

HACKLEY, R. A.: (see Olson, H. F.) (Abstract) 210
HAHN, D. J.: (see Free, E. E.) (Abstract) 66
HALL, HARRY: Sound Analysis, 257–262; (Abstract) 206
HARKER, G. F. HERRENDEN: Principles Underlying the Tuning of Keyboard Instruments to Equal Temperament, 243–256
HAYES, HARVEY: Detection and Location of Laminations in Steel, 220–226; (Abstract) 209
HILL, L. R.: (see Colwell, R. C.) 60–61
HUNT, FREDERICK V.: Apparatus and Technique for Reverberation Measurements, 34–41

J

JONES, ARTHUR TABER: Strike Note of Bells, 199–203
—— Theory of the Haskell Organ Pipe, 196–198; (Abstract) 210

K

KELLER, A. C.: Direct Recording and Reproducing Materials for Disk Recording, 234–242; (Abstract) 206
KLEIN, ELIAS: Spherical Torsion Pendulum as Supersonic Radiation Pressure Meter in Liquids, (Abstract) 210
KOCK, WINSTON E.: Certain Subjective Phenomena Accompanying a Frequency Vibrato, 23–25
—— Vibrating String Considered as an Electrical Transmission Line, 227–233; (Abstract) 210

L

LARSEN, M. J.: (see Lewis, Don) (Abstract) 207
LEWIS, DON: Steady-State and Transient Aspects of Vowel Production, (Abstract) 68
—— Vocal Resonance, 91–99
—— (see Cowan, Milton) (Abstract) 66
—— AND MILTON COWAN: Influence of Intensity on the Pitch of Violin and 'Cello Tones, 20–22
—— AND M. J. LARSEN: Measurement of Timbre, (Abstract) 207
LINDSAY, R. B.: Finite Acoustic Filters, 211–216; (Abstract) 209
—— AND T. G. BARNES: Filtration of Elastic Waves in Solid Rods with Membranes as Side Branches, 42–47
LIVINGSTON, STANLEY D.: Application of Elastic Suspensions to High Speed Newspaper Presses, (Abstract) 66

M

MASSA, FRANK: Loudspeakers for High-Fidelity Large Scale Reproduction of Sound, 126–132; (Abstract) 67
—— (see Olson, Harry F.) 48–52
MCALLISTER, G. STANLEY: Acoustical Planning of Broadcast Studios, (Abstract) 68

MEYER, ERWIN: Reverberation and Absorption, (Abstract) 209
—— Reverberation and Absorption of Sound, 155–161
—— Transmission of Sound and Vibrations in Buildings, (Abstract) 206
MORRICAL, KERON C.: Modified Tube Method for the Measurement of Sound Absorption, 162–171; (Abstract) 67
MORRIS, R. M. AND G. M. NIXON: NBC Studio Design, 81–90; (Abstract) 68
MUNSON, W. A.: (see Steinberg, J. C.) 71–80

N

NEWMAN, E. B.: (see Stevens, S. S.) 185–190; (Abstract) 208
—— AND S. S. STEVENS: Nature and Origin of Aural Harmonics, (Abstract) 208
NIXON, GEORGE M.: (see Morris, Robert M.) 81–90; (Abstract) 68
NORRIS, R. F.: Automotive Quieting, 100–103; (Abstract) 66

O

OLSON, H. F. AND R. A. HACKLEY: Combination Horn and Direct Radiator Loudspeaker, (Abstract) 210
—— AND FRANK MASSA: Compound Horn Loudspeaker, 48–52
OLNEY, BENJAMIN: Method of Eliminating Cavity Resonance, Extending Low Frequency Response and Increasing Acoustic Damping in Cabinet Type Loudspeakers, 104–111; (Abstract) 67

P

POHLMAN, A. G.: Present Status of the Mechanics of Sound Conduction in Its Relation to the Possible Correction of Conduction Deafness, 112–117; (Abstract) 65
PROGRAM OF THE FIFTEENTH MEETING, May 4–5, 1936, Chicago, 65–69; Sixteenth Meeting, October 29–31, 1936, New York, 206–210

R

RETTINGER, MICHAEL: Theory of Sound Absorption of Porous Materials, Flexible and Nonflexible, 53–59
—— Theory of Sound Transmission Through Porous and Nonporous Materials, 172–175
RICE, CHESTER W.: Sonic Marker Beacon for Fog Aviation, 26–29
—— Sonic Locator. An Aid to Fog Navigation, 30–33

S

SKUDRZYK, E. J.: Reviews of Contemporary Papers, 263–264
SMALL, ARNOLD: Harmonic Structure of Violin Tones, (Abstract) 206
SNOW, WILLIAM B.: Change of Pitch with Loudness at Low Frequencies, 14–19
STEINBERG, JOHN C.: Positions of Stimulation in the Cochlea by Pure Tones, 176–180
—— AND M. B. GARDNER: Dependence of Hearing Impairment on Sound Intensity, (Abstract) 207
—— AND W. A. MUNSON: Deviations in the Loudness Judgments of 100 People, 71–80
STEVENS, S. S.: On Hearing by Electrical Stimulation, 191–195; (Abstract) 208
—— (see Newman, E. B.) (Abstract) 208
—— AND H. DAVIS: Psychophysiological Acoustics: Pitch and Loudness, 1–13
——, J. VOLKMANN AND E. B. NEWMAN: Scale for the Measurement of the Psychological Magnitude Pitch, 185–190; (Abstract) 208

T

TATUM, G. R.: Effect of High Intensity Sound on Smokes and Other Aerosols, (Abstract) 210
TIFFIN, JOSEPH: Moving Pictures of the Vocal Cords in Operation, (Abstract) 68
TRIMMER, J. D. AND F. A. FIRESTONE: An Investigation of Subjective Tones by Means of the Steady Tone Phase Effect, (Abstract) 207
TWEEDDALE, J. E.: Measurement and Analysis of Ship Vibration, (Title only) 206

V

VOLKMANN, J.: (see Stevens, S. S.) 185–190; (Abstract) 208

W

WATSON, F. R.: Investigation of the Sound Absorptivity of Portland Cement Concretes, (Abstract) 68
—— Reviews of Contemporary Papers, 264
WATSON, N. A.: Articulation Testing by Bone Conduction, (Abstract) 65
WHEATON, E.: (see Bruderlin, H.) (Abstract) 67
WHITE, WILLIAM BRAID: Musical Instruments and Acoustical Science, (Letter to the Editor) 62–63

THE JOURNAL of the Acoustical Society of America

VOL. 8 NO. 4

APRIL, 1937

Published for the

ACOUSTICAL SOCIETY OF AMERICA

by the

AMERICAN INSTITUTE OF PHYSICS
Incorporated

LANCASTER, PA., AND NEW YORK, N. Y.

ACOUSTICAL SOCIETY OF AMERICA

FOUNDED 1929

OFFICERS

P. E. SABINE, *President*
Riverbank Laboratories
Geneva, Illinois

WALLACE WATERFALL, *Secretary*
The Celotex Corporation
Chicago, Illinois

V. L. CHRISLER, *Vice President*
National Bureau of Standards
Washington, D. C.

G. T. STANTON, *Treasurer*
Electrical Research Products, Inc.
New York, New York

F. R. WATSON, *Editor*
University of Illinois
Urbana, Illinois

EXECUTIVE COUNCIL

C. R. HANNA
Westinghouse Electric & Mfg. Company
East Pittsburgh, Pennsylvania

J. S. PARKINSON
Johns-Manville Corporation
Manville, New Jersey

HARVEY C. HAYES
Naval Research Laboratory
Washington, D. C.

J. C. STEINBERG
Bell Telephone Laboratories, Inc.
New York, New York

A. T. JONES
Smith College
Northampton, Massachusetts

IRVING WOLFF
RCA Victor Company, Inc.
Camden, New Jersey

The officers are also members of the Executive Council

EDITORIAL BOARD

F. R. WATSON, *Editor*

F. A. FIRESTONE
C. R. HANNA
VERN O. KNUDSEN
WALLACE WATERFALL
E. C. WENTE
PAUL E. SABINE, *Ex officio*

Manuscripts for publication should be submitted to F. R. Watson, University of Illinois, Urbana, Illinois.

Proof and all correspondence concerning papers in the process of publication should be addressed to the Publications Manager, American Institute of Physics, 175 Fifth Avenue, New York, New York.

Subscription price to members $4 per year. To nonmembers: Domestic rate $6 per year; Foreign rate $6.60 per year.

Changes of address and complaints of failure to receive the Journal should be addressed to the Publications Manager. New copies can be sent free only if notice of nondelivery is received within three months of date of issue.

The Journal of the Acoustical Society of America is published quarterly at Prince and Lemon Streets, Lancaster, Pennsylvania.

Entered as second-class matter at the Post Office at Lancaster, Pennsylvania.

Copyright, 1937, by the Acoustical Society of America.

THE AMERICAN INSTITUTE OF PHYSICS

INCORPORATED

ADMINISTRATIVE STAFF

HENRY A. BARTON, *Director*

ELMER HUTCHISSON, *Assistant Director*

MADELINE M. MITCHELL, *Publications Manager*

Office: 175 Fifth Avenue, New York, New York

JOURNALS

THE PHYSICAL REVIEW

REVIEWS OF MODERN PHYSICS
JOHN T. TATE, *Editor*
J. W. BUCHTA, *Assistant Editor*

JOURNAL OF APPLIED PHYSICS
ELMER HUTCHISSON, *Editor*

THE JOURNAL OF THE OPTICAL SOCIETY OF AMERICA

THE REVIEW OF SCIENTIFIC INSTRUMENTS
F. K. RICHTMYER, *Editor*

JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA
F. R. WATSON, *Editor*

THE JOURNAL OF CHEMICAL PHYSICS
HAROLD C. UREY, *Editor*

THE AMERICAN PHYSICS TEACHER
DUANE ROLLER, *Editor*

GOVERNING BOARD

For The American Physical Society:

JOHN T. TATE, *Chairman*
University of Minnesota

GEORGE B. PEGRAM, *Secretary*
Columbia University

KARL T. COMPTON
Massachusetts Institute of Technology

For The Optical Society of America:

E. C. CRITTENDEN
National Bureau of Standards

W. B. RAYTON
Bausch & Lomb Optical Company

F. K. RICHTMYER
Cornell University

For The Acoustical Society of America:

HARVEY FLETCHER
Bell Telephone Laboratories

VERN O. KNUDSEN
University of California at Los Angeles

WALLACE WATERFALL
The Celotex Corporation

For The Society of Rheology:

WHEELER P. DAVEY
Pennsylvania State College

MELVIN MOONEY
United States Rubber Products, Inc.

R. L. PEEK, JR.
Bell Telephone Laboratories

For The American Association of Physics Teachers:

HOMER L. DODGE
University of Oklahoma

PAUL E. KLOPSTEG
Central Scientific Company

FREDERIC PALMER, JR.
Haverford College

The American Institute of Physics is a membership corporation of which five societies are the founding members. These five societies are the principal national societies devoted to pure and applied physics in America. Their names, and those of their representatives who form the Governing Board of the Institute, are given above. The principal work of the Institute is the publication of journals sponsored by the societies or by itself. The names and the editors of the journals are also listed above.

The broad purpose of the Institute is to represent, in all matters of wide or common interest, the five thousand or more members and subscribers associated with the Founder Societies and the journals. It aims to advance the science and profession of physics, and to promote cooperation between pure research, the applied sciences and the industries. To achieve these aims, it is greatly in need of an endowment and is empowered by law to administer grants and funds of any kind.

This number of THE JOURNAL OF THE ACOUSTICAL SOCIETY OF AMERICA
was issued April 1, 1937

LANCASTER PRESS, INC., LANCASTER, PA.